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Hearing aid processing of loud speech and noise signals: Consequences for loudness perception and listening comfort.

PhD thesis, July 2006

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Industrial PhD-project at Ørsted●DTU, Acoustic Technology
and Widex A/S, Audiological Research Laboratory, in collaboration
with the Ministry for Sciences, Technology and Innovation

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Enterprise supervisor: Carl Ludvigsen

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Abstract and acknowledgments

English abstract

Hearing aid processing of loud speech and noise signals: Consequences for loudness perception and listening comfort.

Sound processing in hearing aids is determined by the fitting rule. The fitting rule describes how the hearing aid should amplify speech and sounds in the surroundings, such that they become audible again for the hearing impaired person. The general goal is to place all sounds within the hearing aid users' audible range, such that speech intelligibility and listening comfort become as good as possible.

Amplification strategies in hearing aids are in many cases based on empirical research - for example investigations of loudness perception in hearing impaired listeners. Most research has been focused on speech and sounds at medium input-levels (e.g., 60-65 dB SPL). It is well documented that for speech at conversational levels, hearing aid-users prefer the signal to be amplified by approximately half the amount of the hearing loss (in dB). This places the amplified speech signal approximately in the middle of the users' audible range, at a comfortable listening level. However, there has been little research on the optimal gain-prescription for soft and loud sounds. At present, such prescriptions are based mainly on logic, as there is limited evidence on what type of amplification is best for these input-levels.

The focus of the PhD-project has been on hearing aid processing of loud speech and noise signals. Previous research, investigating the preferred listening levels for soft and loud sounds, has found that both normal-hearing and hearing-impaired listeners prefer loud sounds to be closer to the most comfortable loudness-level, than suggested by common non-linear fitting rules.

During this project, two listening experiments were carried out. In the first experiment, hearing aid users listened to loud speech and noise signals with built-in level-variation (62 – 82 dB SPL). The signals had been compressed with seven different compression ratios, in the range from 1:1 to 10:1, yielding different degree of overall level-variation in the processed signals. Subjects rated the signals in regard to perceived level variation, loudness and overall acceptance. In the second experiment, two signals containing speech and noise at 75 dB SPL RMS-level, were compressed with six compression ratios from 1:1 to 10:1 and three release times from 40 ms to 4000 ms. In this experiment, subjects rated the signals in regard to loudness, speech clarity, noisiness and overall acceptance.

Based on the results, a criterion for selecting compression parameters that yield some level-variation in the output signal, while still keeping the overall user-acceptance at a tolerable level, is suggested. It is also discussed how differences in speech and noise components seem to influence listeners ratings of the test signals. General recommendations for a fitting rule, that takes into account the spectral and temporal characteristics of the input signal, is given together with suggestions for further studies. Finally, the experimental methods used for the listening tests in this project are discussed «««.

Danish abstract

Høreapparat processering af kraftige tale og støj signaler: Konsekvenser for lydstyrkeopfattelse og lyttekomfort.

Lydbehandling i høreapparater er bestemt af den tilpasningsregel som er implementeret i apparatet. Tilpasningsreglen beskriver hvordan apparatet skal forstærke tale og andre lyde, sådan at de bliver hørbare igen for den hørehæmmede lytter. Det generelle mål er at placere alle lyde inden for høreapparat-brugerens hørbare område, sådan at taleforståeligheden og lyttekomforten bliver så god som mulig.

Strategier for forstærkning i høreapparater er i mange tilfælde fastlagt på basis af empirisk forskning – fx undersøgelser af lydstyrkeopfattelsen hos hørehæmmede personer. Den meste forskning har været fokuseret på tale og lyd ved medium lydniveauer (dvs. 60-65 dB SPL). Det er veldokumenteret at når det gælder tale ved alm. konversations-niveau, så foretrækker brugerne en forstærkning svarende til omkring det halve af høretabets størrelse (i dB) henover frekvensspektret. Dette placerer talen omtrent i midten af lytterens hørbare område, ved et komfortabelt lytteniveau. Derimod har der været udført meget lidt forskning omkring den optimale forstærkning for svage og kraftige lyde. På nuværende er forstærknings-principper for denne type input baseret på logik, da der mangler beviser for hvilke indstillinger der er bedst for disse lyde.

Dette Ph.d.-projekt har været fokuseret på lydbehandlingen af kraftige tale- og støj-signaler. Tidligere forskning omkring foretrukne lytte-niveauer for svage og kraftige lyde har vist, at både normalhørende og hørehæmmede foretrækker at kraftige lyde placeres tættere ved et komfortabelt lytteniveau, end man ville forvente.

Under dette projekt blev der udført to lytteforsøg. I det første forsøg, lyttede otte høreapparat-brugere til kraftige tale og støj-signaler med indbygget niveau-variation (62 – 82 dB SPL). Signalerne var blevet komprimeret med syv forskellige kompressions-ratioer (1:1 til 10:1), hvilket gav forskellig grad af niveau-variation i de processerede signaler. Forsøgspersonerne vurderede test signalerne mht. oplevet niveau-variation, lydstyrke og overordnet accept af signalet. I det andet forsøg blev to tale signaler, indeholdende tale og støj ved 75 dB SPL RMS-niveau, komprimeret med seks kompressions-ratioer (1:1 til 10:1) og tre udsvingstider (40, 400 og 4000 ms). I dette forsøg vurderede forsøgspersonerne de processerede signaler med hensyn til lydstyrke, talens tydelighed, støj og overordnet accept af signalet.

Baseret på resultaterne foreslås et kriterium for valg af kompressions parametre, som giver en vis niveau-variation i høreapparatet output, samtidig med at den overordnede bruger-accept forbliver på et tåleligt niveau. Det diskuteres også hvordan forskelle i tale og støj komponenter synes at influerer på lytternes oplevelse af test signalerne. Der gives et generelt forslag til en tilpasningsregel, hvor input-signalets spektrale og temporale karakteristika er medtaget i beregningen af høreapparatets forstærkning, og der forslås yderligere forsøg til belysning af dette emne. Endelig diskuteres den eksperimentelle metode anvendt ved lytteforsøgene i dette projekt «««.

Acknowledgments

This industrial PhD-project took place from May 2003 to July 2007, at Ørsted●DTU, Acoustic Technology and the Audiological Research Laboratory at Widex A/S. Institute supervisor was Torben Poulsen (associate professor at Acoustic Technology) and enterprise supervisor was Carl Ludvigsen (leader of the Audiological Research Laboratory). Financial support was provided by Widex A/S and the Ministry for Sciences, Technology and Innovation.

The author would like to thank colleagues at Widex A/S and Acoustic Technology. Special thanks goes to the eight hearing-impaired subjects who participated in experiments during this project, Carsten Paludan-Müller at Widex, who developed the MATLAB-model used for processing the test signals, Jørgen Rasmussen at Acoustic Technology, who provided technical assistance during the listening tests, and Brent Kirkwood and Helen Connor who helped with proofreading of this report.

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1. Introduction

The aim of the present PhD-project has been to investigate hearing aid processing of loud sounds, and to suggest amplification-strategies for this type of input that may be implemented in hearing aid fitting rules.

In everyday life, hearing aid users may be exposed to several different listening environments, depending on their lifestyle, work and leisure interests. A listening environment can be defined as a situation where the listener is placed physically in a given location, listening to the sounds in the immediate surroundings. This location could be a room with people talking at a noisy cocktail party or a quiet open square in the city. The listener may then move on to another location with a different sound environment, for example a quieter room or a busier street with more traffic. Alternatively, the listener may be exposed to different sound environments, while still located in the same place. For instance this would occur in a cinema theatre, where the audience watch and listen to different scenes in a movie.

One attribute of a given listening environment is the overall sound level, measured at the eardrum. Noticeable changes in the sound level may occur when the listener moves from one listening environment to another, or they may occur within the same environment. Also, noticeable level fluctuations between or within environments may occur often (e.g. every ten seconds) or infrequently (e.g. every minute) depending on the situation.

1.1 Level variations in speech and environmental sounds

One example of level-variation is the change in the level and spectrum of speech, which occur as a function of vocal effort. Conversational speech is produced with an average level of 62-65 dB SPL (measured at a distance of 20 cm from the mouth), and the slope of the speech spectrum declines by 3 dB per octave above 500 Hz (Byrne et al, 1994). When changing vocal effort from soft to loud, speech energy in the mid- and high-frequency region is increased more than at lower frequencies. Figure 1.1 shows the average long-term spectra for normal, raised, loud and shouted vocal efforts, as specified in the ANSI-S3.5 standard (1997).

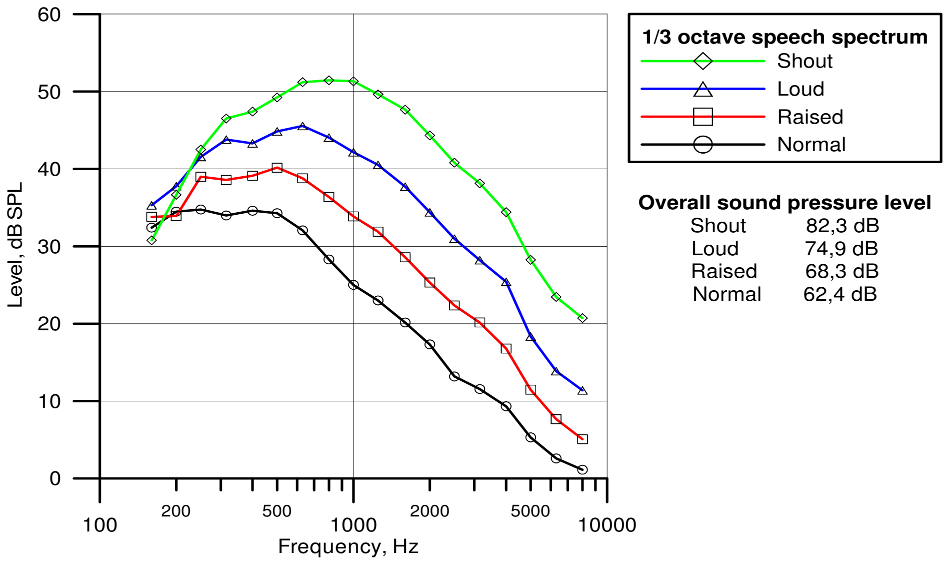

Figure 1.1. Long term 1/3-octave levels for normal, raised, loud and shouted vocal efforts, according to the ANSI-S3.5 standard. Root-Mean-Square levels for each vocal effort are also shown (ANSI, 1997).

Speech is the essential tool of human communication, and prosodic cues such as vocal effort supplements the linguistic message and adds information about the context of the situation and the speaker's mood and intentions. One example is, when two people are having a conversation at a party, and one of them raises his vocal effort in order to be heard (also known as the "Lombard-effect", see Pick et al, 1989).

The spectral changes occurring with increased vocal effort, taken together with the noisy background, are perceived by the listener as if the speaker is trying to overcome the noise. The listener in turn may move closer to the talker to improve signal to noise ratio, making it easier for the talker to get the message across. The listener may also use the information to prepare his or her own vocal effort, when turn-taking takes place (Erber et al, 1998). It therefore seems important that the hearing aid convey information about vocal levels to the hearing-impaired listener.

Apart from speech, the hearing aid user will also be exposed to a variety of environmental sounds. In a study by Wagener et al. (2002) twenty hearing aid users made ear-level recordings of typical listening situations in their daily life. Subjects were instructed to record 5-10 minutes of each situation. The investigators then classified the recordings into three classification groups (with subgroups); **(1)** *conversation without background noise*, **(2)** *conversation with background noise*, (e.g. in a car, bus or café) and **(3)** *other situations* - that is, situations with no conversation, being in a car/bus/train, shopping, reading a newspaper, etc.

The recording material reflected how the overall sound level changes from one environment to another. The lowest mean RMS-level of 56.6 dB SPL occurred in the situation "reading newspaper", and the highest mean level of 91.1 dB SPL was found in the situation "car/bus/train with no conversation taking place" (see figure 1.2). Recordings in the subgroups "conversation in car/bus/train" and "bicycling" were found to have mean RMS-levels beyond 80 dB SPL, while all other recordings (excluding the "reading newspaper"-situation) had levels in the range from 60-75 dB SPL.

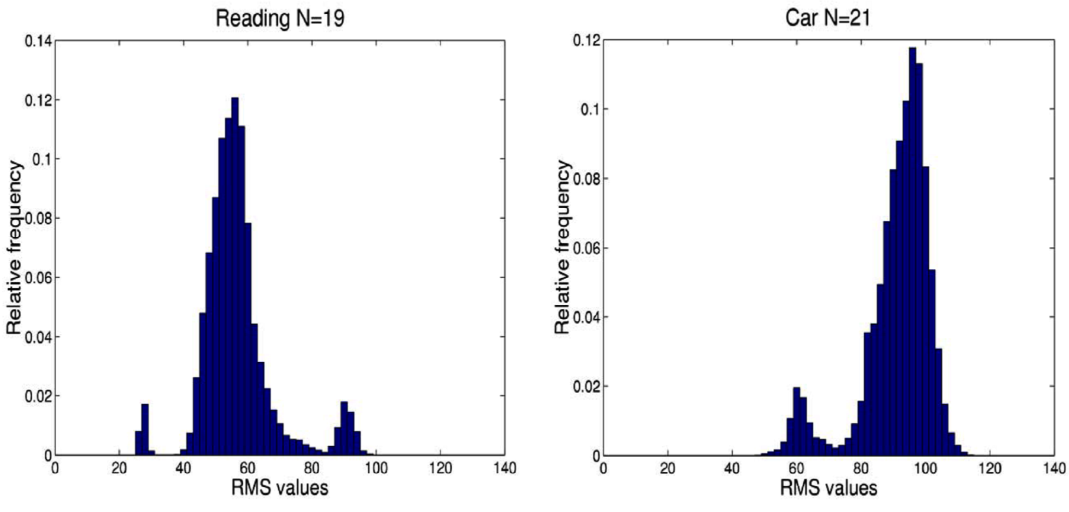

Figure 1.2. To the left, the distribution of RMS-levels in the subgroup "Reading newspaper". This group of recordings had the lowest mean RMS-level of 56.6 dB SPL. The highest mean level of 91.1 dB SPL was found in the subgroup "Car/bus/train", shown to the right (Wagener et al, 2002).

The level corresponding to the 10 % percentile of the level distribution in the "reading newspaper"-group was 46.6 dB SPL. The level corresponding to the 90 % percentile in the “car/bus/train”-group was 101.3 dB SPL. Thus, in this study the dynamic range of the environmental sounds encountered by the listeners spanned across 58 dB, with most sounds having RMS-levels from 60-75 dB SPL and above.

The perceptual effect of speech and environmental sounds over this range of levels will of course depend on the degree of attention they receive. That is, sounds may either be interesting and meaningful to the listener, or they may be irrelevant and even annoying - for example if being too loud or interfering with speech during a conversation.

In some cases the overall sound level, as well as changes in the level, may provide useful information and receive part of the listener’s attention. Combined with spectral information, the sound level may act as an auditory cue, helping the listener to identify sound sources in the surroundings and the action taking place. In addition to this, input from other senses, such as vision, will also help to complete the picture. The situation described earlier with a conversation in a noisy room is one example of this. Another situation is that in a busy street, listening to vehicles approaching or moving away. Here it may be vital for the listener to perceive level differences in order to navigate safely through traffic.

1.2 Level variations in reproduced sound

Apart from naturally occurring level-changes, hearing aid users will also be exposed to varying sound levels when listening to electrically reproduced sound. For instance, this would occur when watching TV, being at a movie theatre or listening to a car radio. In these situations, speech and environmental sound may originally have been produced at medium sound levels (for instance a speaker, talking at a normal vocal effort). When the signal is reproduced, the presentation level may be lower or higher, compared to the original recording level.

In some cases, presentation levels of reproduced sounds tend to be higher compared to the levels of the same sounds in a natural environment. One example is the overall level in movie theatres which has been found to be quite high, e.g. from 70-80 dB(A) (Salo, 2000). Even higher levels beyond 80 dB(A) have been reported, occurring especially in trailers and commercials (BSI, 1999). Figure 1.3 shows the level variations in dB(A), measured in the middle of a theatre during a movie. The overall level for the whole movie was 79 dB LA_{eq}.

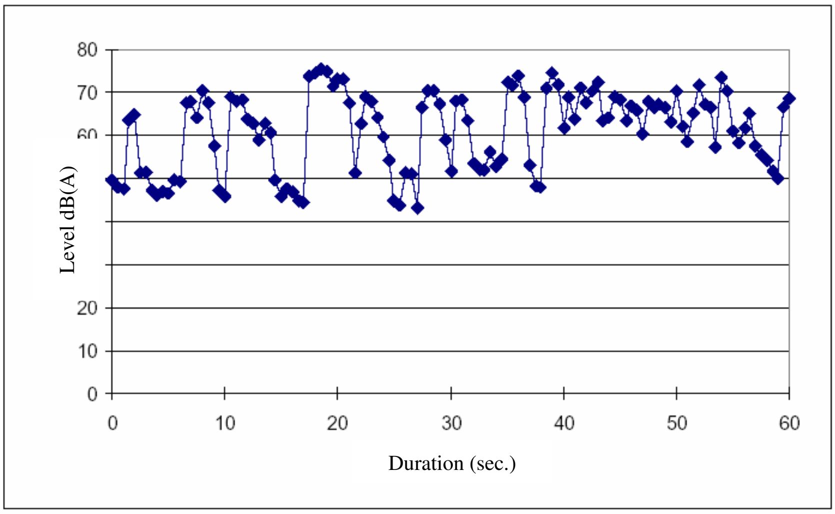

Figure 1.3. Level variation (dBA) during one minute of the movie “Gone in 60 seconds”. Measurement was made in the Tennispalatsi Theater in Helsinki. The level meter was positioned in the middle of the theatre and the measurement was done with a 500 ms integration time (Salo, 2000).

Similarly, when listening to a car-radio, passengers tend to adjust the volume relative to the noise-level in the car. The noise level in car compartments depends on the combined noises from the road, wheels, wind and engine, as well as the isolation materials used for dampening these sounds.

A typical scenario is when driving speed changes from slow to fast. In order to keep listening to speech or music from the radio, the driver turns up the volume to achieve better audibility of the signal. Even though the level of the radio may be quite high, this usually does not bother passengers because the signal is partially masked by the car noise. Only when the speed drops again, the radio suddenly appears too loud and the volume is turned down.

Figure 1.4 shows spectra for the noise in a car cabin and the frequency response of the car radio (pink noise) measured at the driver's seat (Lydolf, 2003). It is seen how the lower frequency parts of the radio signal receive a poorer signal-to-noise ratio, compared to higher frequencies - and more so at the higher driving speed.

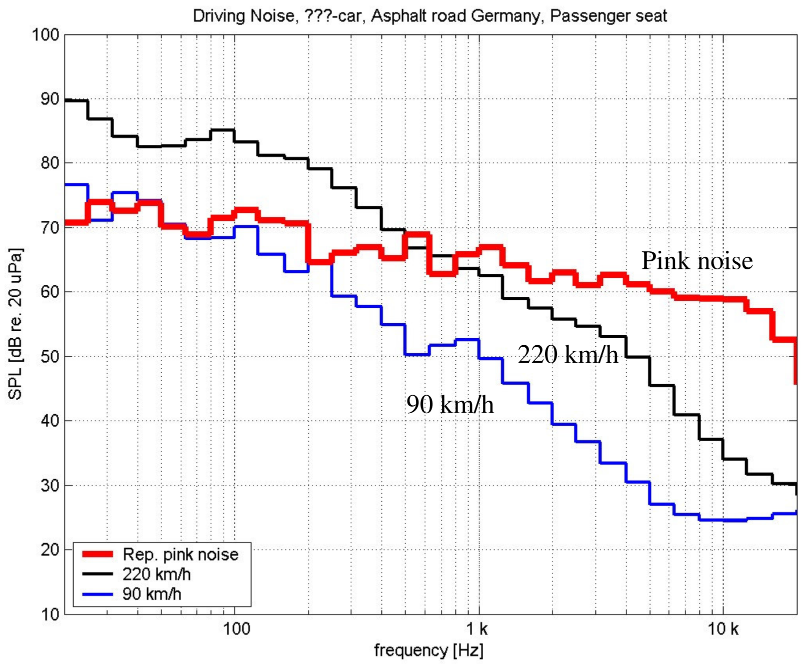

Figure 1.4. 1/3-octave levels of driving noise at 90 and 220 km/h, versus frequency response of the audio system. Measurement was done in a German manufactured car, at the passenger seat with a ¼ inch B&K microphone and PULSE measurement-system (Lydolf, 2003).

One special issue with reproduced sound is that the source signal may already be altered, before being presented to normal-hearing listeners. This is the case with many radio broadcasts, where the speaker's voice is compressed to compensate for variations in his or her vocal level. Similarly, soundtracks for movies may be compressed in different ways and the signal split into several loudspeakers with speech only coming from a centre speaker behind the screen. Another example is the car radio, which may be connected to the engine such that presentation-level and spectrum are altered at high driving speeds.

1.3 Hearing aid processing of the dynamic input-range

Soft, medium and loud sounds are perceived within a listener's auditory range. The normal auditory range across frequency is shown in figure 1.5. In the figure, the lower border of the range (the *hearing threshold*) is represented by the minimal audible field and the upper border (the *upper comfortable loudness level*) is represented by the 120 phon equal loudness contour. In the 1 kHz region, the range from the *hearing threshold* to the *upper comfortable loudness*-level (UCL) is about a 120 dB, depending on the test methods used for establishing the levels.

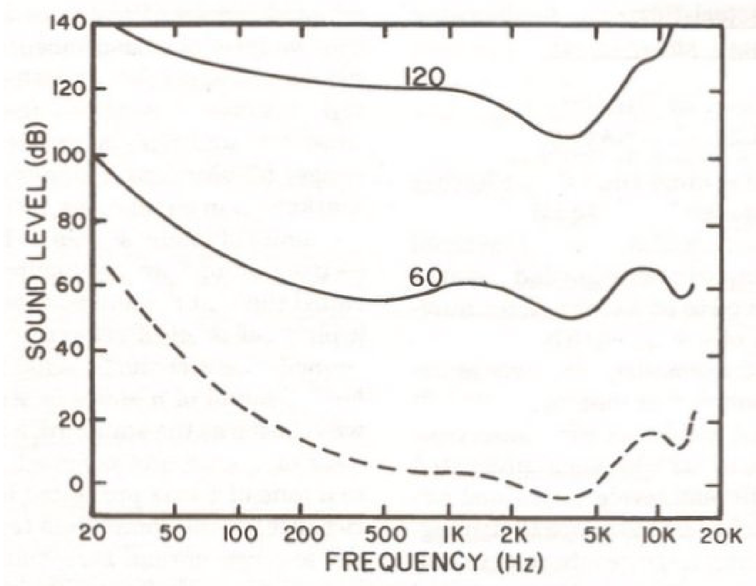

Figure 1.5. An illustration of the normal auditory range. The *hearing threshold* is represented by the minimal audible field (dotted line). The *most comfortable loudness level* is represented by the 60 phon loudness contour and the *upper comfortable loudness level* by the 120 phon loudness contour (modified from Robinson & Dadson, 1956).

The auditory range may be divided into a lower and an upper part, separated by the *most comfortable loudness*-level (MCL). The MCL is the level (or range of levels), at which the listener perceives the presented sound as having a comfortable loudness. In normal-hearing listeners the average level of the MCL has been found to coincide with the 60 phon equal loudness contour for pure tones (Christen & Byrne, 1980).

In a hearing-impaired listener with a sensorineural loss (i.e., of cochlear origin), the lower and upper border of the auditory range is raised compared to normal, although the UCL is not raised to the same degree as the threshold. The level of the MCL is also raised, but stays slightly above the middle (5-10 dB) of the restricted auditory range (Pascoe, 1988). In addition, the impaired listener's perception of loudness growth is altered compared to the growth perceived by normal-hearing persons.

The challenge for the hearing aid manufacturer, and the audiologist in the field, is to select and fit the hearing aid such that sounds in the normal auditory range become audible to the hearing-impaired listener. In the era of linear amplification, this was primarily a question of two parameters: The selection of a gain-frequency response that placed conversational speech at the listener's MCL, and a suitable setting of the *maximum power output* (MPO) to avoid loudness discomfort from the hearing aid. The user then would adjust the volume control of the device, to make soft sounds more audible or reduce loud sounds if they appeared too loud.

With the introduction of non-linear gain in the 1980'ies, it became possible to better address some of the psychoacoustic attributes in sensorineural hearing loss. The reduced sensitivity for soft sounds and the abnormal loudness growth at supra-threshold levels could then be compensated for by applying high gain for low input levels and gradually decreasing the gain as the input level increases (like is done in *Wide Dynamic Range Compression*, WDRC). In addition, the implementation of multiple compression channels has made it possible to attempt an imitation of the processing taking place in human auditory filters.

Several fitting rationales for prescribing non-linear gain have been developed. They include both generic rationales, intended for all brands of hearing aids on the market (Cornelisse et al, 1995; Valente & Van Vliet, 1997; Byrne et al, 2001), and device-specific rationales developed by individual hearing aid manufacturers. The latter are often developed to suit specific features and the technology used in a given hearing aid series.

In non-linear hearing aids, the gain-frequency response for a given input-level will depend on the input-output characteristics specified in the fitting algorithm. In particular, the interaction between the dynamic aspects of the compressor (i.e., the setting of compression ratio and time constants) and the temporal and spectral properties of the input signal, will govern the degree of gain applied at a given input level. In a digital hearing aid, apart from the compression system, several subsystems will operate simultaneously to reduce the negative effects of noise, to increase focus on the source signal (speech) and to adapt the sound processing to different listening environments.

Today, the objective of most fitting algorithms is that speech should be audible and intelligible under various conditions. Also, soft sounds should be made audible to the listener and loud sounds should be presented without causing discomfort. A more general objective could be that the user should be satisfied with his hearing aid in all listening situations, without the need of adjusting any controls on the device.

In other words, the hearing aid should be as "transparent" and "adaptive" as possible when it comes to providing audibility, good speech intelligibility and listening comfort for various sounds. This trend is also seen in the physical dimensions of modern hearing aids, as many manufactures put emphasis on making their devices as small and light as possible, and to provide so-called "open fittings" (with large vent-sizes) to reduce occlusion effects and pressure-sensations in the ear canal.

1.4 Focus area in this report

Even though validation studies have been carried out concerning the user benefit of non-linear fitting rules, it is still not entirely clear what amplification strategies should be used for sounds in the lower and higher parts of the input-range. Most research on the optimal gain and compression settings have been focusing on speech and sound at medium input levels (e.g., 60-65 dB SPL). Also, some non-linear fitting rationales have partly been built on knowledge gained in the development of earlier linear rules that were focused on the amplification of conversational speech levels.

In contrast, very little research has been done on the non-linear sound processing of soft and loud sounds (a review is given in chapter 3). Some authors have noted the lack of investigations in this area, among them Byrne et al. (2001) who stated…

"*There remains the question of deriving prescriptions for inputs that are significantly higher or lower than average. For present, such prescriptions must be based mainly on logic as there is very limited evidence on which compression thresholds and ratios are best.*"

This PhD-project has been focused on the hearing aid processing of loud speech and noise signals. As noted earlier, many daily listening situations contain sound energy in the range above the most comfortable listening level in a normal-hearing person. In the data collected by Wagener et al. (2002), the total range of sound levels corresponded to 58 dB, and most of the recorded mean levels were found to be above 60 dB SPL. It is therefore of great relevance to investigate hearing aid processing of sound in the upper part of the input-range.

The hearing aid fitting rationale should be able to manage loud input sounds whenever they occur, such that they do not cause any discomfort. On one hand, it seems appropriate that the hearing aid user is presented with some level variation in the output signal, such that high input levels are perceived as being louder than medium input levels. On the other hand, if loud sounds are not handled well by the hearing aid, they may cause discomfort to the hearing aid user. The user may turn down the volume control of the device if possible or even turn it off completely. Turning down the volume too much may compromise audibility.

In regard to the non-linear processing of loud sounds, the selection of the *maximum power output* is of less importance, because the gain is reduced towards zero at the high end of the device's input-range. At high input levels, the output level from the device will become equal to the level of the direct sound incidence, reaching the eardrum through the ear mould and ventilation tube. The setting of the MPO (or the *output compression limiter)*, will "only" be important to avoid audible distortion, when the maximum output level of the device is reached.

Of greater importance are the input/output characteristics specified by the fitting algorithm across frequency. The degree of gain and compression for loud sounds, and the attack and release times used in the compressor, may affect the listeners' impression of the loudness, speech intelligibility, noisiness, listening comfort and sound quality, as well as level cues providing information to the listener about the present auditory environment.

Several aspects of the relationship between gain and compression settings and the subjective impression of the processed sound could be investigated. In this project, two experiments were carried out, with a group of experienced hearing aid users. The focus in the first experiment was on the degree of level variation in the hearing aid output, which listeners can accept for loud speech and noise signals. This experiment was carried out in continuation of earlier studies showing a preference for presenting all input levels close to the most comfortable loudness level of the hearing aid user. The second experiment was focused on the perceptual effect of the time constants used in the compressor. Earlier studies investigating this aspect have primarily used speech and noise signals at medium input levels. In the second experiment, loud speech and noise were processed with varying release times and compression ratios, to see the perceptual effect of compression with such signals.

This report is divided into seven chapters. Chapter 2 provides some theoretical background on the perception and measurement of loudness in hearing-impaired listeners. Chapter 3 focuses on technical aspects of linear and non-linear amplification, and reviews earlier studies that investigated the perceptual effects of compression on signals at various input-levels. The 4^{th} and 5^{th} chapters describe the two experiments carried out during this project. And in chapter 6 and 7, the results of the experiments are discussed and it is considered if a general fitting-rule can be built on the basis of the findings. Suggestions for future work are also given.

1.5 Units in the objective and subjective description of sound

In connection with the quantitative and qualitative description of sound throughout this report, certain units should be categorized in relation to the stage in the human hearing system from where they arise. The measures of sound along this pathway can be described by a filter model with three stages (Pedersen & Fog, 1998). A slightly modified version of this model is suggested here (shown in figure 1.6).

The first stage of the model contains the physical sound, which is quantified at the measurement point M1 (equivalent to the entrance of the ear canal). The units of measurements at M1 include the instantaneous *sound pressure level* (dB SPL), the long term *root mean square level* (RMS-level) as well as representations of the sound's spectral and temporal characteristics.

Filter 1 is equivalent to the sense of hearing (i.e. the ear canal, middle ear, inner ear and the auditory pathways) and marks the border to the second stage. The second stage contains the perceptual measurements of the sound made by the listener himself, and quantified at the measurement point M2. The measurements made at M2 include the psychoacoustic metrics of *hearing threshold level* (HTL), the *loudness level* (equal loudness contours), the *most comfortable loudness level* (MCL), the *upper comfortable loudness level* (UCL) and the *loudness* (Sones), as well as the *intelligibility of speech* tested via speech audiometry.

Filter 2 marks the border to the third stage, and resembles the non-acoustical (or psychological) factors of auditory perception - such as the preference and expectations of the listener, in combination with the context of the listening situation. In the third stage, the listener makes subjective judgements about the sound quality (such as *clarity*, *sharpness* or *fullness*) and the intelligibility of speech. He may also decide whether he likes or dislikes the sound present in the given situation. These judgements are quantified at the point M3, for instance by having the listener indicate which one of two listening programs he prefer or can accept (preference testing) or via rating on categorical scales.

The measurements made at M1 will depend on the processing taking place in the hearing aid, as well as the acoustic coupling between the device and the ear canal. The measurements made at M2 and M3 will depend on the condition of the second and third filters. If the sensitivity and selectivity of the hearing sense is reduced, this will affect the perceptive measurements at M2. Also, the affective measurements made at M3 will depend on the prominence and the weighting of the non-acoustical factors in the second filter ««.

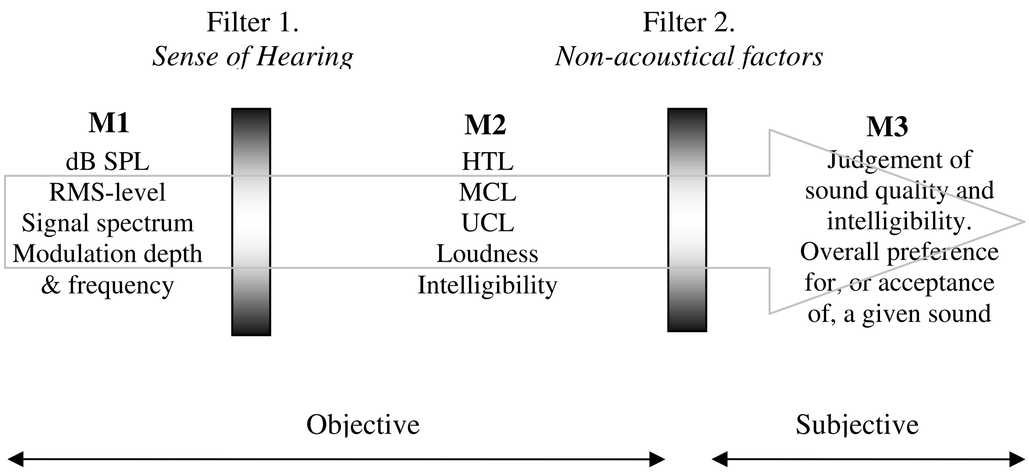

Figure 1.6. Filter model for the quantitative and qualitative description of sound in the human hearing system (own illustration, based on Pedersen & Fog, 1998).

2. Perception and measurement of loudness in hearing-impaired listeners

The focus in this chapter will be on the effects of sensorineural hearing loss on loudness perception. Loudness is a relevant parameter in regard to gain-prescription in non-linear hearing aids. The loudness sensation produced by the hearing aid output will affect listening comfort and the users' overall impression and acceptance of the device. Objectives for the loudness delivered by the aid, either based on empirical or theoretical findings, should therefore be included in the hearing aid fitting rationale.

Sensorineural hearing loss may have many different origins, such as acoustic trauma, age-related changes, hereditary predisposition, ototoxicity, hypoxia or inner-ear diseases like Morbus Ménière, etc. (for a review see Pickles, 1988). The individual types of loss may affect auditory perception in different ways, although there still is a lack of reliable clinical measures to detect subtle differences between them. However, there is general agreement that most sensorineural losses affect the mechano-electrical transduction processes in the inner and outer haircells on the Organ of Corti.

2.1 Cochlear damage and its effects on the auditory response area

The outer hair cells (OHC) are believed to be responsible for the non-linear phenomena seen in the normal mammalian cochlea. Figure 2.1 shows velocity-intensity functions recorded in a chinchilla, before and in four time-intervals after an injection of furosemide (Robles & Rich, 1991). Furosemide is a diuretic, known to cause disruption of outer hair cell activity. The tone-stimulus was a 9000 Hz tone-pip presented at nine levels in the range from 20 to 100 dB SPL. Recordings were done at two locations on the basilar membrane, corresponding to the 9000 Hz and 1000 Hz regions.

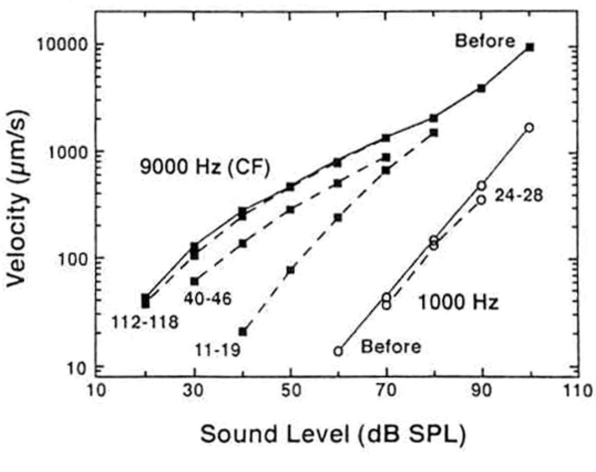

Figure 2.1. Velocity-intensity functions recorded in a chinchilla at locations corresponding to 1000 Hz and 9000 Hz. Functions were recorded before and in four time-intervals after an injection of furosemide (Robles & Rich, 1991).

The function recorded at the 9000 Hz location, before the injection, is non-linear. That is, the vibrations are amplified and compressed in the range from 30-90 dB SPL. Right after the injection (11-19 sec), the gain at the lower sound levels is reduced and the function becomes almost linear. The reduction in the response-magnitude is about 25-30 dB. Only at high sound levels (80 dB SPL) the function is similar to the normal condition. Later (40-46 sec and 112-

118 sec) the function starts to regain its normal shape, as the effect of the furosemide is diminishing. In contrast, no effect of the drug is seen in the function recorded at the 1000 Hz location. At this location, the function is quasi linear before and after (24-48 sec) the injection.

There have been other findings of cochlear nonlinearity; including two-tone suppression (Ruggero et al. 1992), combination-tone generation (Harris et al, 1989) and the presence of measurable otoacoustic emissions (Kemp, 1978). The absence of these phenomena, in combination with electroscopic inspection of missing outer hair cells, imply that active "motor" processes in the OHC are responsible for the high sensitivity and frequency selectivity of the human ear.

The theoretical function of the active process is shown in figure 2.2. The transduction in the OHC is believed to reduce the friction on the basilar membrane at the place of the characteristic frequency. Thereby, the peak of the passive travelling wave becomes sharpened. This may lead to an increased excitation of the inner hair cells, which are the "sensory" cells firing to the auditory nerve (de Boer, 1983).The effect of the active mechanism is high at low sound levels, but gradually reduces as the input level increases. At high input levels the response becomes more and more linear, indicating that passive forces driving the membrane movements are dominant.

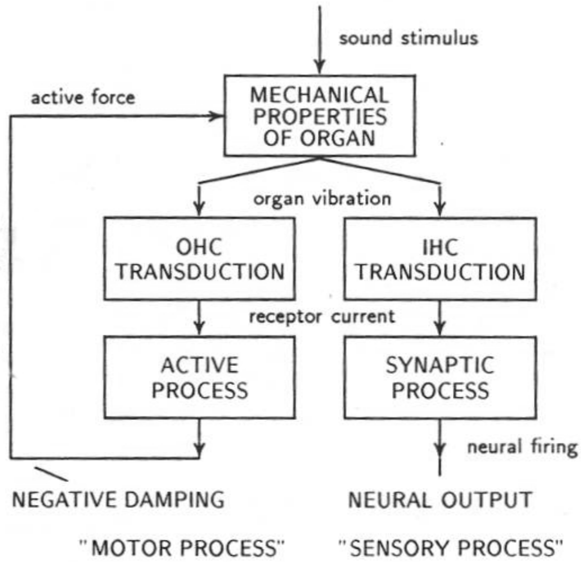

Figure 2.2. Schematic illustration of the motor and sensory processes in outer and inner hair cells (illustration from Launer, 1995).

Damage to the active process alters auditory processing in a number of ways: elevation of hearing thresholds, abnormal loudness growth, changes in auditory filter bandwidth, reduced frequency selectivity and reduced temporal processing (Moore, 1996). These adverse effects influence listeners' perceptions and impressions of sounds, as well as the intelligibility of speech - especially when encountered in noisy surroundings.

Abnormal loudness growth and the elevation of hearing thresholds can be demonstrated by observing changes in the equal loudness level contours. Figure 2.3 shows the average shape of the contours in young normal-hearing listeners, according to the ISO 226-standard (ISO, 2003). These curves were obtained from measurements in the free field, using a loudness matching procedure with pure tones as stimuli and a 1000 Hz tone as the reference.

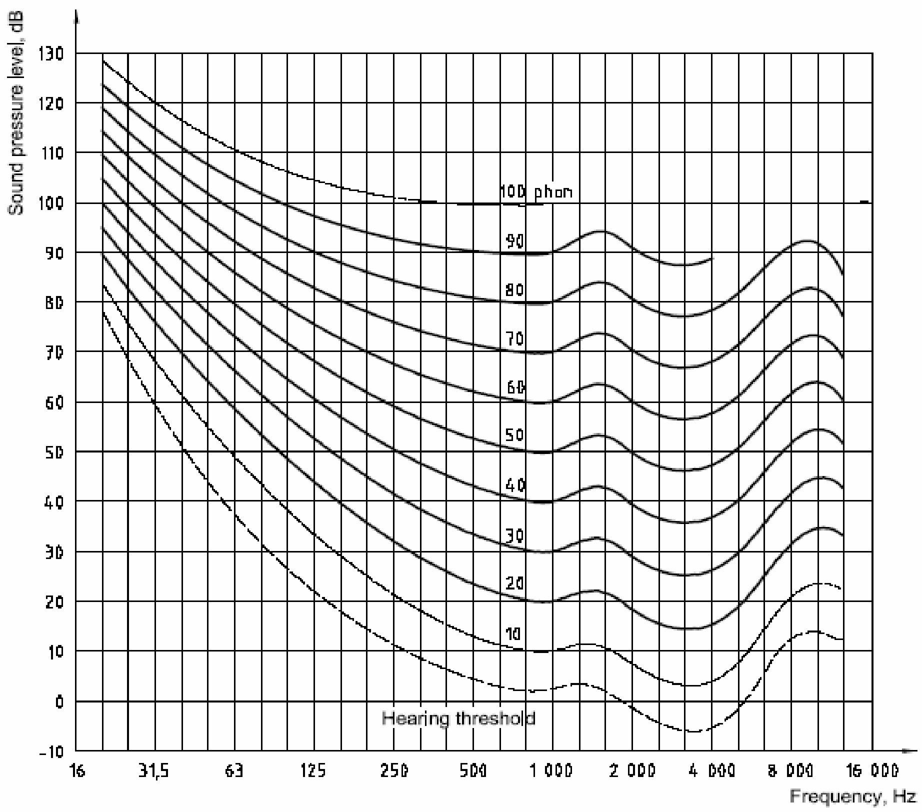

Figure 2.3. Normal equal-loudness level contours for pure tone stimuli, referenced to a 1000 Hz tone. Binaural free-field listening, frontal incidence (ISO, 2003).

Measurements of equal loudness contours are problematic to carry out in hearing-impaired listeners. One method is to present the reference tone (to which the loudness at other frequencies is matched) in a region with normal-hearing thresholds. This method was used by Barfoed (1975), who matched loudness at seven frequencies to a reference tone at 500 Hz, in subjects with high frequency hearing loss. Equal loudness level contours made from measurements with two subjects are shown in figure 2.4.

It can be seen that the change in hearing threshold modifies the level contours, compared to their normal shape in figure 2.3. In both subjects, the lower contours have been compressed, particularly in the region with more pronounced hearing loss. The decibel required to match a 10 dB change in the 500 Hz tone, e.g. going from the 30 phon to the 40 phon level, is much smaller at higher frequencies than at lower frequencies.

In contrast to this, the spacing between the upper contours (80-100 phon) is relatively even across frequency and similar to the one in the ISO 226-standard. The raised threshold curve and the abnormal narrow spacing of the lower loudness contours are related to the loss of the active mechanism in the cochlea.

From Barfoed's data, it can be inferred that variability exists in the shape of contours among hearing-impaired subjects. This was even the case in subjects with relatively similar audiogram configurations (not shown here). For the hearing-impaired listeners, some of the variability may be related to the type of damage in the cochlea. Also, part of the variability may be caused by measurement difficulties. For instance, the variability has been shown to increase, as the separation between test and reference frequency increases (Poulton, 1989).

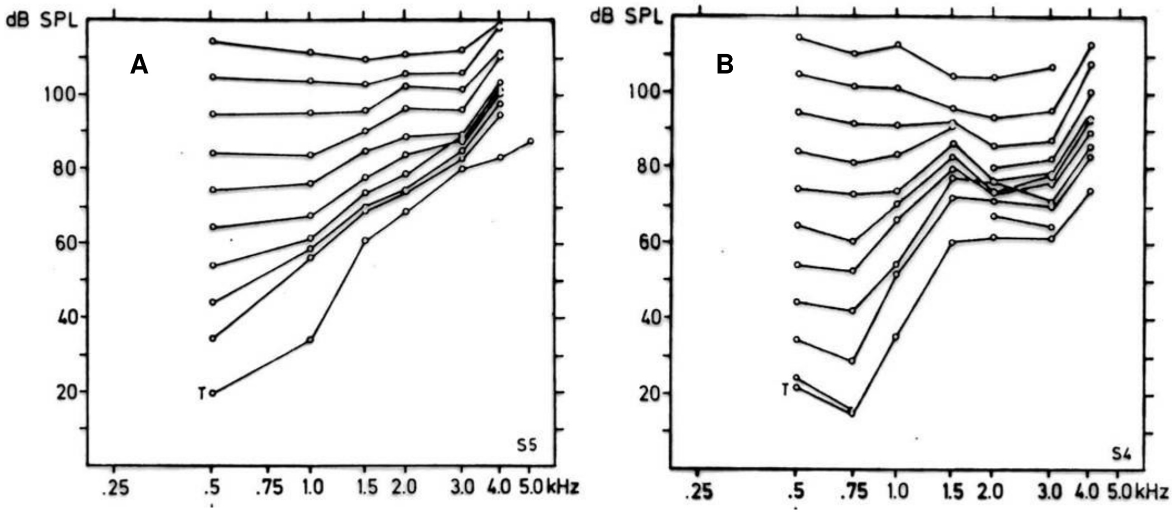

Figure 2.4 Equal loudness contours measured monaurally in two subjects with sensorineural hearing loss. Subject A (left) with a moderately sloping loss and subject B (right) with a steeper loss in the 1 kHz-region. Sound levels are calculated from voltage at headphone terminals (Barfoed, 1975).

Thus, equal loudness level contours are distorted in hearing-impaired listeners with sensorineural hearing loss – especially at sensation levels right above threshold. And even for similar losses, they may be distorted in different ways. It should be noted, that the data behind the average loudness level contours for normal-hearing listeners (as shown in figure 2.3), also show great variability. Thus, variability in loudness-level for the same hearing threshold is not only associated with hearing impairment, but also with normal-hearing (Elberling, 1999).

2.2 Subjective loudness measured by magnitude estimation

The equal loudness level contours can only be considered as an indirect measure of loudness sensation. Also, the loudness level is only valid for the pure-tone or narrow band noise used as stimulus. A more direct measure, which relates the physical magnitude of sound to its subjectively perceived loudness, can be made with a scaling procedure. In this type of measurement, the listener's sensation of loudness is transformed into a different domain, where it is represented, for example, by a number or by a marking on a visual scale.

In one scaling procedure, the *absolute magnitude estimation*, the listener assigns a number on an open scale to describe the loudness of the presented signal. This procedure was originally proposed by Stevens & Davis (1938), who assumed that normal-hearing listeners judge loudness on a ratio-scale. They measured growth functions for loudness and a number of other senses like smelling and temperature. Based on this data, Stevens & Davis derived a function that describes the relationship between sound pressure level and the perceived strength of loudness. For sound pressure levels beyond 40 dB SPL, this is a straight line on a log-log scale that can be described by a *power function*:

$$N = kp^{0.6}$$

where N is the loudness in Sones, p is the sound pressure in micropascals (µPa) and k is a constant that determines the numerical scaling on the abscissa. For sound pressure levels below 40 dB SPL the measured loudness function exhibits a steeper growth than at higher levels. The power function was modified to describe this phenomenon:

$$N = k(p-p_0)^{0.6}$$

where p_0 is the hearing threshold in dB sound pressure level. The subtraction of p_0 has greatest influence at low sensation levels. The unit *Sone* was chosen to represent the sensation of loudness, where 1 Sone is arbitrarily defined as the loudness of a 1 kHz pure tone presented at a sensation level (SL) of 40 dB above the hearing threshold (equal to 40 phon). It follows that a 1 kHz tone having a loudness of 2 sones (equal to 50 dB SL) is perceived as being twice as loud as the same tone having a loudness of 1 Sone (equal to 40 dB SL). Figure 2.5 shows the loudness function in normal-hearing listeners estimated by magnitude estimation, compared to the function estimated by the power law by Stevens & Davis.

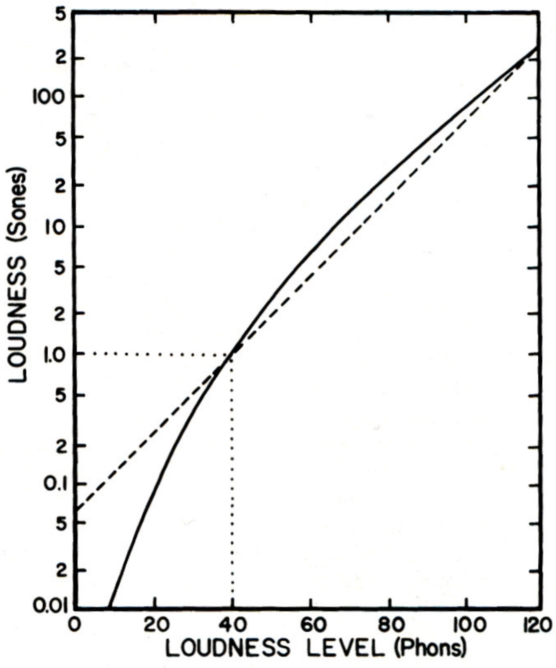

Figure 2.5. Relationship between loudness (Sones) and loudness level (Phons) of a 1-kHz tone. The broken line is the prediction of loudness made by the original power law by Stevens & Davis (1938) (illustration from Durrant & Lovrinic, 1995).

A variation of magnitude estimation is *magnitude production*, where the listener is given a number and then asked to adjust the level of the stimulus so that the perceived loudness matches that number (Stevens, 1957). In another variation, *restricted magnitude estimation*, the listener is given a restricted range (e.g. 1-100) and asked to choose a number within this range when judging the loudness (Geller & Margiolis, 1984).

2.3 Loudness growth in hearing-impaired listeners

The shape of the loudness growth function is altered in people with sensorineural hearing loss. A traditional description of impaired loudness growth is shown in figure 2.6 (the thin dotted line, data by Fowler, 1937). The impaired function exhibits a much steeper slope than the normal function, at sensation levels just above the raised threshold.

At higher sensation levels, the slope of the function becomes gradually less steep and finally coincides with the average normal function. This is known as the *loudness recruitment*-phenomena. Like the abnormal narrow spacing between the level contours in figure 2.4, this is related to the loss of the compressive mechanism in the cochlea. Note also the relation of the impaired function to the linearized velocity-intensity functions in figure 2.1.

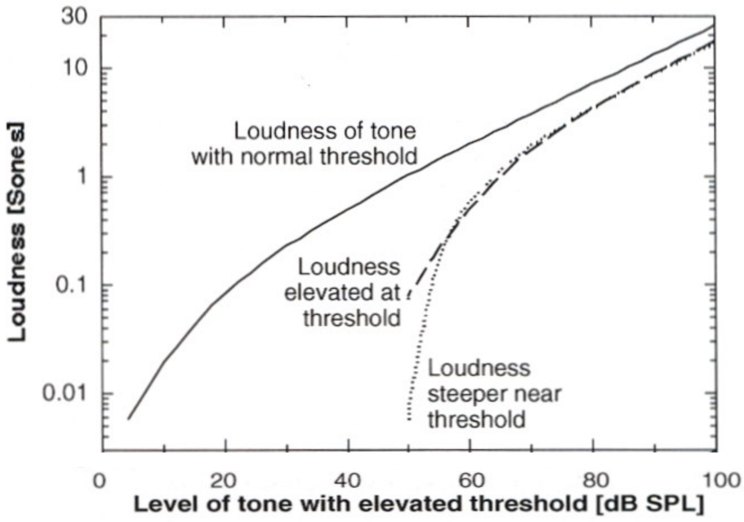

Figure 2.6. Loudness growth in normal hearing listeners (solid line) and in a hearing-impaired listener (thin dotted line). The function made with a dashed line represents a case of softness imperception (Florentine and Buus, 2001).

Different types of recruitments have been described in the literature. Some research has been concerned with the lower part of the impaired loudness function. Florentine and Buus (2001) suggested that the loudness perceived by listeners with cochlear loss is not as soft at threshold, as it is for normal-hearing listeners. Instead, the hearing-impaired exhibit an inability to perceive sounds presented at threshold as being soft. Therefore the loudness function near threshold may not be as steep as in the traditional description of recruitment, rather it resembles the growth at threshold in normal listeners. This phenomenon has been denoted *softness imperception* (shown by the dashed line in figure 2.6).

There may also be variability in the degree of recruitment at high sensation levels. Five different types of recruitment have been defined based on investigations, using the alternating binaural loudness-balance procedure in patients with unilateral losses (Jerger, 1962; Brunt, 1994). The five types are shown in figure 2.7. The dotted line in the figure represents equal loudness between the normal and impaired ear.

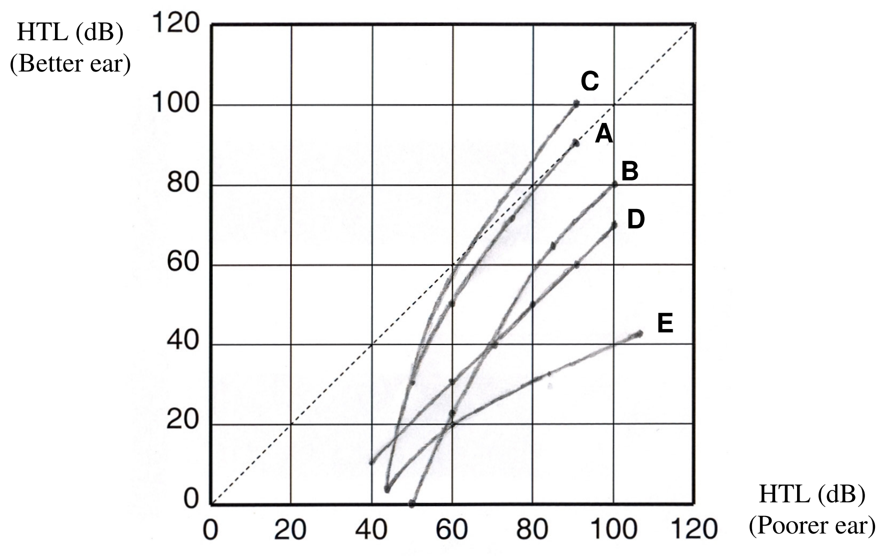

Fig. 2.7. Comparison of five types of loudness recruitment reported in the literature. (A) Complete recruitment, (B) partial recruitment, (C) over-recruitment, (D) no recruitment and (E) decruitment. Dotted line represents equal loudness between the normal and impaired ear (own illustration based on Jerger, 1962 and Brunt 1994).

Complete recruitment (A) is defined by the case where the impaired ear's function coincides with the normal ear's function at higher sensation levels (+/- 10 dB). The case where equal loudness judgments are made at equal sensation levels in both ears (+/- 10 dB), is defined as no recruitment (D). In partial recruitment (B), the shape of the impaired function is comparable to the one seen in complete recruitment, but the function never reaches normal perceived loudness at high presentation levels. It is also defined as a function that falls midway between complete and no recruitment. Over-recruitment (C) is seen when the impaired function exceeds the normal function by more than 10 dB at high presentation levels. Thus, in this case the impaired ear perceives the stimulus to be louder than the normal ear, at the same presentation level.

Decruitment (E) can be defined as the opposite of recruitment, where the slope of the impaired function is shallower (compressed) compared to the normal function. The opposite is the case with recruitment, where the slope is steeper (expanded) relative to the normal function.

Complete, Partial and over-recruitment are typically seen in sensorineural hearing losses. The difference between these functions may be related to the type of cochlear loss, and the complexity of witch the outer and inner hair cells are affected by damage. No recruitment and decruitment are usually not seen in sensorineural losses, but have been associated with conductive and retrocohlear losses (Thomsen et al, 1981).

Thus, for the same degree of hearing loss, individual differences in loudness growth may exist. This has implications for the fitting of non-linear hearing aids. If the gain target for a loud input sound is based on average data for complete recruitment, an individual listener with partial recruitment may perceive the loudness as being lower than estimated by the fitting rationale. In that case, fine tuning of the fitting may be needed. Alternatively, the fitting could also be based on individual loudness measurements, obtained via categorical scaling methods (discussed later).

2.4 Loudness summation in normal and hearing-impaired listeners.

Monaural loudness not only depends on the intensity level of the sound, but also on the bandwidth of the stimulus in relation to width of the auditory filters. The bandwidth and shape of these filters has been investigated by several researchers (e.g. Fletcher, 1940; Zwicker and Scharf 1965; Moore et al, 1990).

In normal-hearing listeners, the frequency range of hearing is spanned by approximately 25 filters or *critical bands*, which correspond to actual intervals on the basilar membrane of approximately 1.25 mm. The filter bandwidths in Hz increase with frequency and are approximately equal to 1/3 octave intervals. At low sound levels, filters have a rounded exponential shape. At higher levels, the shape becomes asymmetric towards the low frequency side of the filter.

In the case where components of a complex sound fall within the same critical band, the total loudness is a function of the total loudness level of the complex - which in turn reflects the total sound pressure level. When the same components fall in different critical bands, then the total loudness approaches the sum of the individual loudnesses of the components.

The effect of the critical band on loudness perception has been shown in classical experiments, using loudness comparison (e.g., Zwicker & Feldtkeller, 1967; Scharf & Houtsma, 1995). When the total energy in a noise is kept the same, but the bandwidth of the noise is increased, then the perceived loudness level increases as the bandwidth exceeds a given size (the critical band). This phenomenon is denoted *loudness summation.* The degree of loudness summation is found to be greatest at medium presentation levels, and less pronounced at low and very high presentation levels.

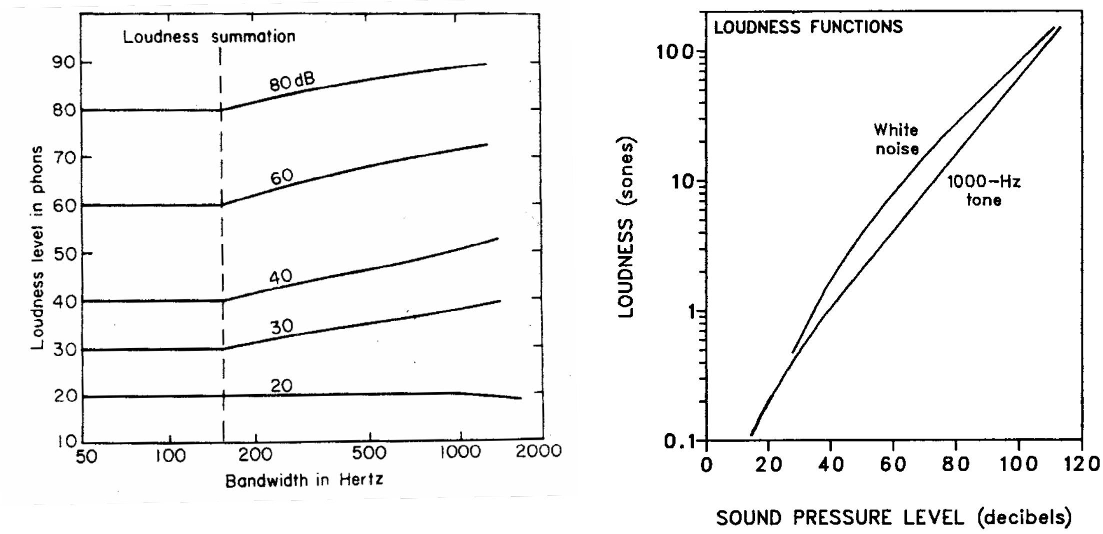

Figure 2.8. Two demonstrations of loudness summation. Left, the measurement of critical bandwidth by the means of an increase in loudness as a function of noise bandwidth (Zwicker & Feldtkeller, 1967). To the right, Loudness functions for white noise and for a 1000 Hz pure tone (Scharf & Houtsma, 1995).

In hearing-impaired listeners with sensorineural losses, loudness summation has been found to be less pronounced or absent (Scharf & Hellman, 1966; Bonding & Elberling, 1980, Florentine & Zwicker, 1979). The reason for the absence of loudness summation in listeners with sensorineural impairment, may partly be due to increased bandwidth of the auditive filters. Filters have been shown to become broadened in sensorineural losses, and it has been shown theoretically that the excitation within each band is reduced (specifically in the centre of the bands). This yields an overall lower specific loudness per band, which results in reduced summation (Moore, 1996). The broadening of auditive filters is believed to be indirectly caused by the loss of the compressive mechanism in the cochlea.

Moore also notes that the reduced summation of loudness means that listeners with impaired loudness require larger changes in presentation level to achieve the same loudness for broadband signals as normal listeners (e.g. when trying to normalise loudness with a non-linear hearing aid). The advantage of this reduction is that there will be less difference in the estimation of the most comfortable and uncomfortable levels, estimated with narrowband and broadband signals respectively. This is relevant in relation to clinical scaling methods used for hearing aid fitting, as the use of narrowband signals will no longer underestimate loudness for broader signals, like speech. This also applies to the usage of normative data for MCL and UCL-levels used as basis for non-linear gain-prescription.

2.5 Subjective loudness measured by categorical scaling

In *magnitude estimation* of loudness, the naive listener makes a freely and non-biased judgement of the perceived loudness. The assigned numbers are made on an absolute scale of loudness, and a function for loudness growth in a log-log plot can be obtained by averaging values assigned by the test group.

An alternative method is the *categorical loudness scaling*, which has also been used as a clinical tool in conjunction with hearing aid fitting procedures. This method is based on the assumption that, in real life, listeners assign a verbal label to the loudness of a sound. In a typical scaling procedure, the listener is asked to select between a fixed number of verbal categories. The main categories may be labelled as "very soft", "soft", "comfortable", "loud" and "very loud". Also, the two categories "inaudible" and "too loud" are often used to mark the lower and upper limits of the scale. Two examples of categorical loudness scales are shown in figure 2.9.

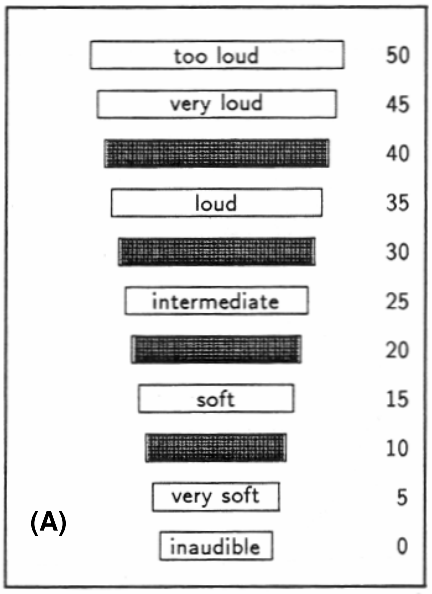

Figure 2.9. Two examples of categorical loudness scales. In (A) the touch-screen with 11 categories used by Launer (1995). In (B) the interval-scale with five main categories used by Gabrielsson et al (1985). In that scale, subjects mark the perceived loudness with a pencil or pen.

Loudness functions obtained with magnitude estimation and categorical scaling respectively cannot be directly compared, as they differ in procedure and shape of the functions (Stevens, 1975). On a log-log scale, functions obtained with categorical scaling have a more concave-down shape compared the linear shape of functions obtained with magnitude estimation (figure 2.10).

At high sensation levels, the deviation between functions is due to the fact that the scale used in magnitude estimation contains no upper limit. That is, listeners are allowed to use very high numbers to describe the loudness of loud sounds. In contrast to this, a categorical scale has a fixed upper limit, which acts as a "roof" of listeners' rating of high presentation levels. This influence the rating towards the upper end of the scale, making the slope of the function shallower compared to magnitude estimation (Stevens & Galanter, 1957).

Categorical loudness scaling made on scales with spacing between categories has also been found to yield smaller standard deviations compared to magnitude estimation, magnitude production and restricted magnitude estimation. This has been explained by the presence of fixed verbal categories on the scale. These act as fix points, making listeners bias their response toward the main categories instead of putting their response in-between (Heller, 1991).

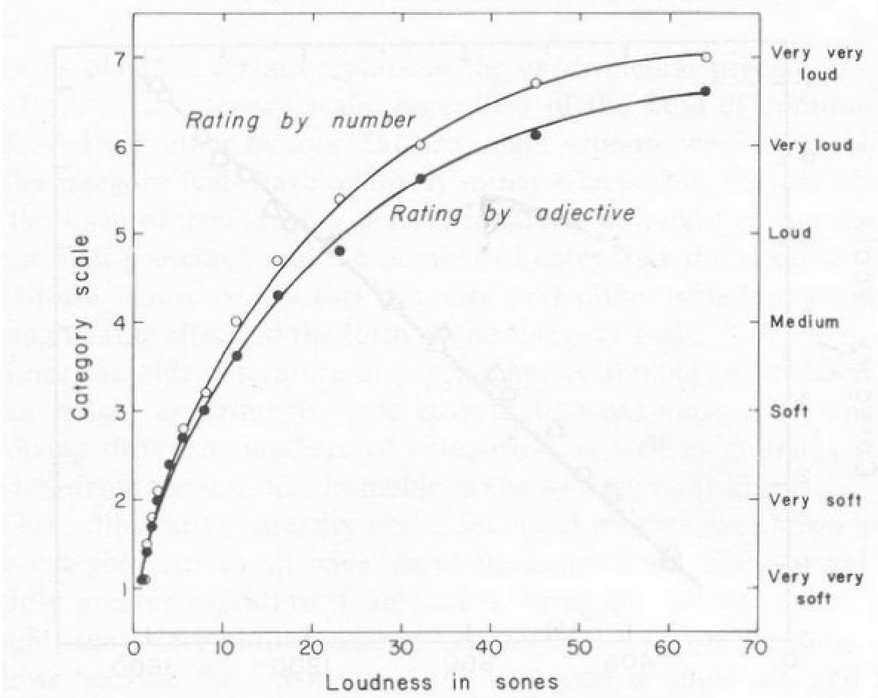

Figure 2.10. Differences in the shape of loudness functions, obtained with categorical scaling (black dots) and magnitude estimation (white dots). The stimulus is a white noise signal, ranging from 40 dB SPL to 100 dB SPL, in steps of 5 dB. The function obtained by categorical scaling exhibits a more curvilinear shape relative to the ratio scale of loudness (Stevens & Galanter, 1957).

In a study by Launer (1995), three scaling methods were compared: *Absolute magnitude estimation* made on an open scale, *restricted magnitude estimation* made on a restricted scale from 0-50 and *categorical scaling* using seven verbal categories for loudness (as shown in figure 2.9A).

Subjects were presented with narrowband noise with a bandwidth of 200 Hz, centred around 1 kHz. Each scale was evaluated two times with overlapping presentation levels. First, 21 stimuli were presented randomly in the range from 0-60 dB HL. Secondly, same number of stimuli were presented in the range from 30-90 dB HL. In figure 2.11, the function obtained with categorical scaling is compared to the functions obtained with absolute magnitude estimation (left) and with restricted magnitude estimation (right).

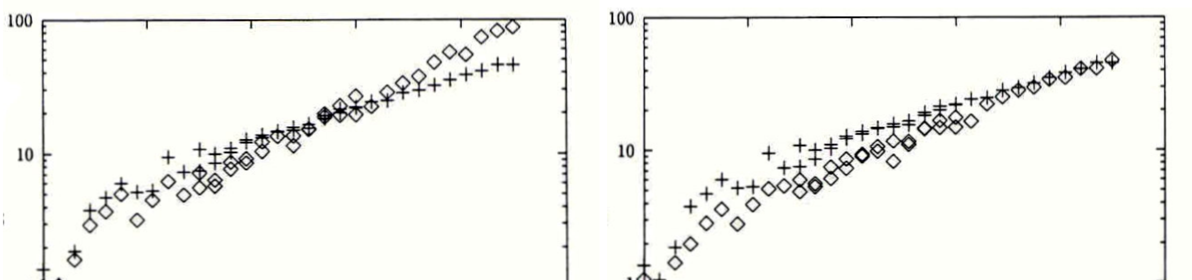

Figure 2.11. Left: Comparison of loudness functions obtained with Categorical Scaling (+) and Absolute Magnitude Estimation (◊). Right: Comparison of Categorical Scaling (+) and Restricted Magnitude Estimation (◊) (Launer, 1995).

The overall shape of the functions is the same for the three methods, but as expected there is a deviation at high presentation levels between the categorical and open magnitude scales (left graph). This deviation is not seen in the comparison between the categorical and restricted magnitude scales (right graph). The author concludes that a restricted magnitude scale with many categories (fifty, in this case) and a categorical scale with few categories, yields equal shapes of the loudness functions. Therefore a categorical scale with fewer categories may be just as reliable to use, as absolute or restricted magnitude estimations.

One finding of this study, which is relevant for the experiments in this project, is that splitting the loudness scale into two partly overlapping level-ranges, did not have an adverse effect on the shape of the total functions obtained. When presenting test signals only in a narrow level range, there might be a risk that the subject stretches the scale, such that all categories are employed. But as seen in figure 2.11, the values obtained in the 0-60 and 30-90 dB HL-ranges coincide in the overlapping region from 30-60 dB HL. Stretching of the scale may be avoided either by presenting stimuli that covers the whole dynamic range or by carefully instructing the listener by presenting reference-stimuli at the lower and upper boundaries of the level range, before the scaling procedure begins. Launer noted that stretching of the loudness scale did not appear in his study because the absolute magnitude estimation was carried out first. This may have provided listeners with a reference of the total range of presentation levels used in the test.

The loudness scale used by Launer (1995) had non-labelled categories placed in-between the main categories, i.e. representing intermediate loudnesses between categories. With this type of scale, the listener is given the possibility of graduating his response, if for example the perceived loudness lies in between “comfortable” and “loud”. In another type used by Gabrielsson & Sjögren (1979), the main categories are placed on an interval-scale with 10 major marks, divided into 100 minor marks (shown in figure 2.9B). In this case the listener is asked to use the whole scale, also the intervals between categories, to mark his sensation of loudness.

The rationale for using a categorical scale with intervals is that the listener is made aware of the perceptual spacing between categories. In both the scales shown in figure 2.9, the proximity of “Very soft” to “Inaudible/Min”, as well as “Very loud” being close to “Too loud/Max”, makes the listener critically distinguish which category is perceived in each end of the scale.

In summary, there are some advantages of using categorical scales, compared to magnitude estimation, in subjective loudness estimation. In magnitude estimation, the listener is presented with the abstract task of describing loudness with a number. It has been hypothesized that age, educational background and degree of hearing loss may influence the listener’s ability to perform this task (Studebaker & Scherbecoe, 1988). The use of verbal categories for loudness provides a more natural starting point for the measurement. It is easier for the tester to provide the subject with instructions, and it may also be easier for the subjects to relate the loudness of a signal to a verbal expression (Pascoe, 1978).

In addition, loudness has been found to be one of the attributes that listeners use in their judgement of sound quality. This was investigated by Gabrielsson & Sjögren (1974, 1975, 1977), who asked hearing-impaired listeners to rate the sound quality of different hearing aids. They used 50 different scales as the one in figure 2.9B, each with its own adjective for describing the sound quality. Through factor analysis they found that seven adjectives were significant for the description of sound quality: *loudness*, *clarity*, *fullness*, *spaciousness*, *brightness*, *softness* and *nearness*. These adjectives have been used for later studies on impaired listeners' assessment of non-linear hearing aids (Neuman et al, 1998; Hohmann & Kollmeier, 1995. See also chapter 3).

2.6 The use of categorical loudness scaling for non-linear hearing aid fitting

As illustrated in figure 2.4 and 2.7, the relationship between degree of hearing loss and loudness growth at individual frequencies may not be a simple one. Factors other than the configuration of the loss (e.g. related to the type of cochlear damage) may affect the shape of the functions. If the fitting objective is to normalize loudness perception, then a transfer function that relates normal loudness to the loudness perceived by the user is needed.

Categorical loudness scaling has been implemented as part of fitting rationales for non-linear hearing aids. By performing an individual loudness scaling in the clinic, and using these data as reference for the gain prescription, it was hoped for that objective of normal loudness would be achieved.

The first loudness scaling procedure available for clinical use, was the *Loudness Growth in Octave Bands* (LGOB) by Pluvinage (1989) and Allen et al. (1990). In this procedure ½-octave bands of noise, centred at 250, 500, 1000, 2000 and 4000 Hz are presented at 15 different levels. Subjects then rate the loudness on a categorical scale using six verbal categories, ranging from "very soft" to "too loud".

An example of loudness functions measured in one subject with a severe high frequency loss is shown in figure 2.12. The plots show the relationship between SPL for a given loudness category in the hearing-impaired subject (abscissa) and the SPL for the same category obtained with normal-hearing listeners (ordinate). The average normal function at each frequency (the straight line) is also shown for comparison. The triangles represent the levels that received ratings of "ok" – that is, equal to comfortable loudness or midway between "soft" and "loud".

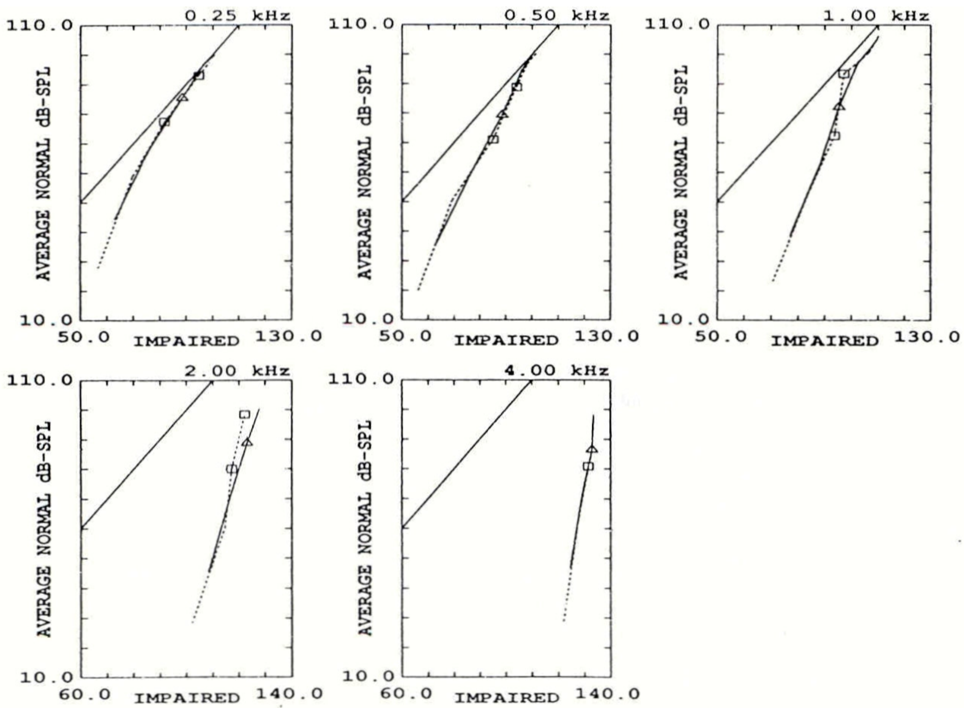

Figure 2.12. Loudness functions obtained with ½-octave bands of noise at five different frequencies and fifteen presentation levels. Triangles represent the presentation levels that received a rating of "ok" (Allen et al, 1990).

The graphs illustrate how loudness growth changes across frequency, for a particular subject. At 250 Hz, the impaired function lies close to the average normal function. As the hearing threshold is raised at the higher frequencies, the impaired function diverges more and more from the normal function, and it becomes steeper. At 1, 2 and 4 kHz, the function never reaches the normal function at high presentation levels - at least not at the presentation levels tested in this subject. Thus, the loudness scaling shows that the occurrence of *complete recruitment* and *partial recruitment* is frequency dependent in this particular listener.

In order to normalize loudness in this subject, a high amount of gain should be applied at low sensation levels. As the slope of the impaired function gradually becomes less step at higher levels, the gain should be reduced and eventually be 0 dB at high levels, when and if the two functions coincide with each other.

Several other loudness scaling procedures were developed during the 1990'ies, (e.g. Kiessling et al. 1993; Hohmann & Kollmeier, 1995; Launer, 1995; Ricketts & Bentler, 1996, Cox et al. 1997). The procedures were developed either as independent procedures, as part of hearing aid test systems, or as an integrated part of commercial hearing aid fitting software. The procedures vary from each other in many aspects, including the instructions given to the subject, the response method, the number of categories used on the scale, the type of stimuli, the range and the spacing between presentation levels.

Also, some of the procedures (e.g. the *Contour Test* by Cox et al. 1997) bias the overall order of presentation levels, such that lower levels are presented first followed by higher levels. The rationale for this paradigm is to raise the listener's acceptance for high presentation levels, and thereby exploring the upper limits of tolerable loudness. Other procedures use a random order of presentation levels.

It has been noted that differences between loudness procedures, as well as variability in the way individual procedures are administered, may undermine the effectiveness of the loudness scaling as a tool for hearing aid fitting. For instance, Jenstad et al. (1997), using normal-hearing listeners, found that the shape of the loudness function is dependent on the chosen order of presentation levels.

Compared to a random order of levels, the functions were shallower when a high-level reference signal was presented at the start of each trial, or when a given level stimulus was preceded by a stimulus having a greater level. This means that loudness functions obtained with random and sequential presentation order cannot be directly compared. Also, normative data for a given scaling procedure obtained with normal-hearing subjects is necessary, in order to apply the data in an individual hearing aid fitting.

Elberling (1999) also showed, that different scaling procedures relate the verbal loudness categories differently to sound level. For instance, for the category "comfortable" a variability in presentation level of 25 dB was found between procedures (figure 2.13). Elberling also investigated the variation in loudness functions obtained with same procedures. In this case, a variability of 35 dB in the presentation level yielding the response "comfortable" was found among subjects. Finally he showed that based on normative data, the loudness function can be estimated in 70-75 % of hearing-impaired listeners, with an accuracy within +/- 5 dB across presentation levels.

This relationship speaks against the use of individual loudness measures, as a basis for increasing the accuracy in non-linear gain-prescriptions.

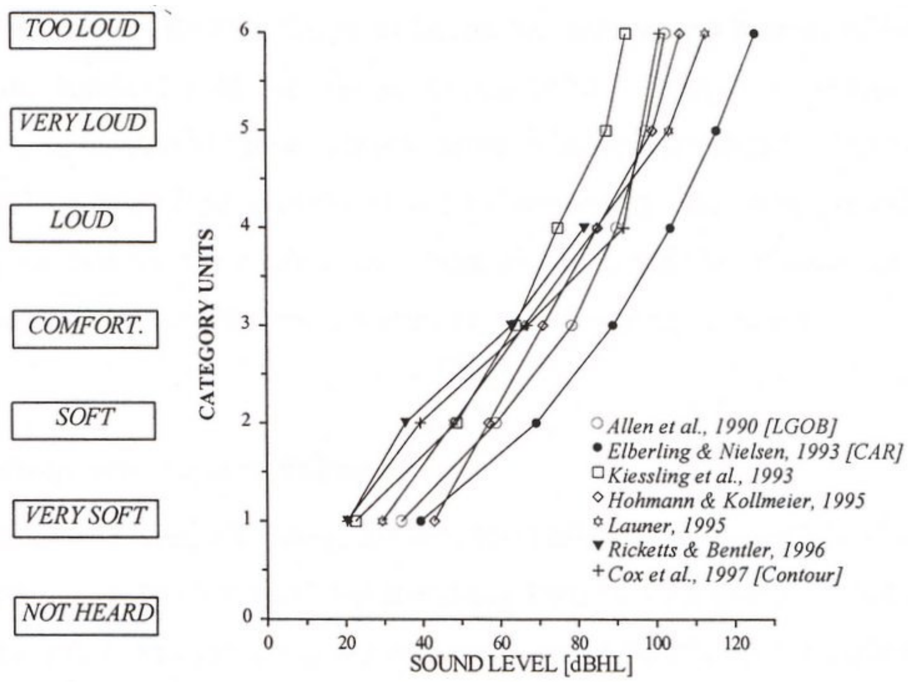

Figure 2.13. A normative study, comparing the shape of loudness function obtained with seven different loudness scaling procedures. (Elberling, 1999).

The issues discussed above have promoted the use of *normative loudness data* in hearing aid fitting – that is, prescription of gain based on normative data describing the relationship between hearing threshold and supra threshold levels collected in a larger population. In one type of normative data, the *most comfortable loudness* (MCL) and *uncomfortable loudness* (UCL) levels are related to the degree of hearing loss (Pascoe, 1986, 1988; Schwartz et al, 1988). The data from Pascoe (1988) is shown in figure 2.14.

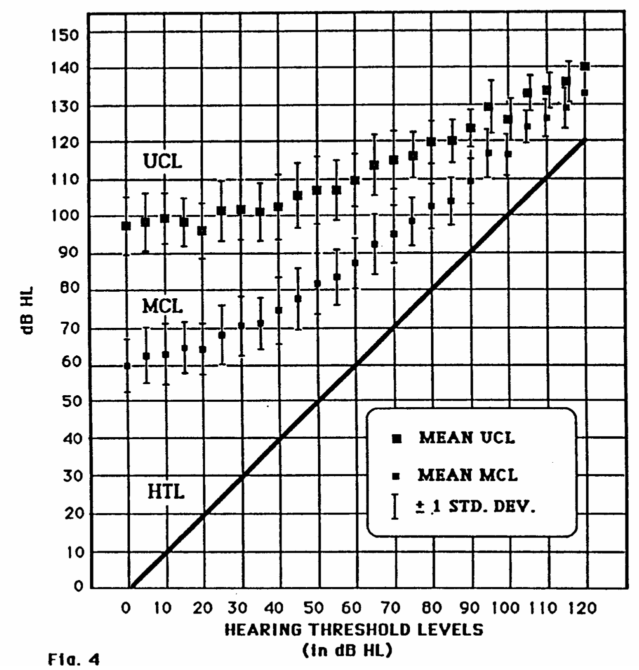

Figure 2.14. Mean MCL and UCL-values from 508 ears are shown. Values were obtained with pulsed tones under headphones and data were collected at 500, 1000, 2000 and 4000 Hz. Because of no significant frequency effect, the data were pooled together. The pooled data are shown above, as well as the +/- 1 standard deviation (Pascoe, 1988).

This data was obtained from 508 ears with hearing threshold levels ranging from 0-120 dB HL. Pascoe used pulsed tones monaurally as stimuli, and asked subjects to select one of ten categories that described their sensation of the loudness. The ten categories were divided into four color-ranges: White (*nothing heard*), yellow (*too soft*, *very soft* and *soft*), orange (*ok soft*, *ok* and *ok loud*) and red (*loud*, *very loud* and *too loud*). Subjects were asked to think of speech at normal vocal level, and use this as a reference for their judgement of the loudness of the tone. For each individual subject, the most comfortable level was defined as hearing level half way between the average level of *ok soft* and *ok loud*. The uncomfortable level was defined as the hearing level where the rating of too *loud* was repeated.

The slopes of the MCL and UCL functions are roughly divided in two sections. From 0 – 45 dB HL the MCL-value increases about 3.3 dB for every 10 dB increase in hearing threshold. Above 45 dB HL the increase is steeper, 7.5 dB per 10 dB HL. The UCL-value stays close to 110 dB HL at losses below 45 dB HL. For losses above 45 dB HL, the UCL-level increases by 5 dB for every 10 dB increase in hearing level. Thus, the dynamic range becomes narrower as a function of threshold. And the most comfortable loudness level is always positioned in the middle of this range (more precisely, 5-10 dB above the middle).

Even though most of the ratings were within 10 dB of the mean values, some individuals deviated as much as 47.5 dB from the mean. Pascoe notes that for 1/3 of the population tested, this variability could result in an over or underestimation of their MCL and UCL levels. This has implications for the use of normative data in hearing aid fitting, especially concerning the need for fine tuning after the initial fitting.

2.6 Summary and implications for hearing aid fitting

This chapter has dealt with the effect of cochlear damage on loudness perception, and the estimation of loudness growth using scaling-techniques and normative data. Two effects of cochlear damage have been considered; *loudness recruitment* and the *absence of loudness summation*. In a hearing-impaired listener, the loudness function is typically steeper close to threshold than in normal-hearing listeners, but it then becomes overlapping with the normal function at higher sensation levels. Different degrees of recruitment exist, but they may not be predictable from the pure tone audiogram alone. This has implications for the fitting of non-linear hearing aids. If the gain target for a loud input sound is based on average data for complete recruitment, a listener with partial recruitment may perceive the loudness as being lower than was the intention. In contrast to this, a listener with over-recruitment may perceive the loudness to be greater than intended.

In the case of loudness summation, broadband versus narrowband signals have been shown to produce less or no change in the perceived loudness in hearing-impaired listeners. This is presumably due to the broadening of auditory filters. This has implications for the effects of the various test signals used in loudness scaling procedures, as well as for the gain prescription for broadband and narrowband signals in hearing aids. For instance, if the goal is to normalize loudness for all sounds, additional gain might be needed to restore normal loudness for a broadband signal, if loudness summation is absent in a given listener.

Categorical loudness scaling has been found to be equally reliable compared to magnitude estimation procedures. Categorical scaling procedures are more time efficient and use fewer and more descriptive categories for describing the loudness. Therefore, this scaling technique has been widely used as a clinical tool for estimating individual loudness growth, often in combination with the fitting of non-linear hearing aids. But due to the variability between procedures and the inter and intra-subject variability observed with same procedures, the use of categorical loudness scaling as a tool for individual hearing aid fitting has been questioned (Elberling, 1999). If the loudness function can be predicted in 70-75 % of the cases, it seems clear that normative loudness data can provide a starting point for the fitting, which is just as good as or better than individual loudness scaling. Today, most hearing manufactures have abandoned the use of individual loudness measures and have started using normative data, as for instance the data by Pascoe (1988) relating MCL and UCL to HTL.

Thus, even though categorical loudness scaling has been abandoned for prescribing hearing aids, the technique is still relevant for obtaining normative data of the perceptual effects of hearing loss – data, which can be used as basis for the development of fitting rationales. Also, categorical scaling can be used to compare the subjective perception of different hearing aid settings. In the following chapter, some studies that applied categorical scaling techniques to assess the subjective impressions of hearing aid compression settings will be reviewed. Categorical scales were also used in the experiments described in this report, to investigate the acceptable level-variation of loud speech and noise signals, processed by a non-linear hearing aid «««.

3. Hearing aid processing of the dynamic input range and earlier investigations on the preferred listening levels for soft and loud input signals

3.1 General principles for gain prescription in non-linear hearing aids

Sensorineural hearing loss causes reduced audibility for soft sounds and abnormal growth of perceived loudness for soft, medium and loud input signals. The main goal for a hearing aid fitting rationale is to amplify speech to audible levels and to optimise speech intelligibility. But besides the speech signal, the hearing aid should also amplify environmental sounds to such levels, that the listener is provided with information about his or her auditory environment. Both speech and environmental sounds are presented at various input levels to the hearing aid microphone, and may still be equally meaningful to the listener.

During the historical development of hearing aids, several principles for gain selection have been proposed. These principles have often been linked to the technology at the time - e.g., the amount of gain available and the capabilities for automatically regulating the gain in the hearing aid. Today there is still no agreement among researchers what should be considered the “best” fitting formula. Also, it might not be desirable to follow just one approach. However, there is general agreement that all fitting rationales should make soft meaningful sounds audible and make loud sounds comfortable and undistorted (Kuk & Ludvigsen, 1999).

In continuation of this, and from a sound quality-point of view, it makes sense to provide the hearing aid user with sensations of the sound level variations in the environment – i.e., there should be some degree of level variation in the output from the hearing aid, reflecting the overall level fluctuations in the surroundings (Dillon, 2001).

3.1.1 Gain prescription for medium input-levels

Until the 1980’s, amplification in hearing aids was generally linear. The signal would receive the same amount of gain, independent of the input level. The hearing aid user would adjust the volume control of the device to some overall desired listening level.

Several generic fitting rules for linear amplification were developed in the 1970’s and 80’s (Berger, 1976; McCandless & Lyregaard, 1983; Seewald et al, 1985; Byrne & Dillon, 1986). Using different approaches, most of these rules prescribed gain to be approximately half the amount of the hearing loss (in dB) across frequencies. This principle was originally proposed by Lybarger (1944).

The half-gain approach was based mainly on empirical findings, showing that hearing-impaired listeners with mild and moderate losses prefer this setting for speech, produced at normal vocal effort (60-65 dB SPL). Applying gain equal to half the loss in dB, would place the speech spectrum close to the most comfortable listening level - approximately in the middle of the listener’s dynamic range.

Insertion gain responses prescribed for a moderately sloping hearing loss by six different linear fitting rules are shown in figure 3.1. Note the diversity in the amount of insertion gain at the higher frequencies. This difference is caused by divergence in the underlying fitting objectives among rules – objectives which may be based on either empirical or theoretical findings. For instance, the goal of the NAL-procedure (Byrne & Dillon, 1986) is to amplify all parts of the speech spectrum to the MCL, such that they contribute equally to its loudness and are equally intelligible (denoted as *loudness equalisation*, see subsection 4.4.3). Another example is the POGO-rule (McCandless & Lyregaard, 1983), which is a straight forward implementation of the half-gain principle, but including a gain reduction at the lower frequencies to avoid upward spread of masking resulting from ambient noise.

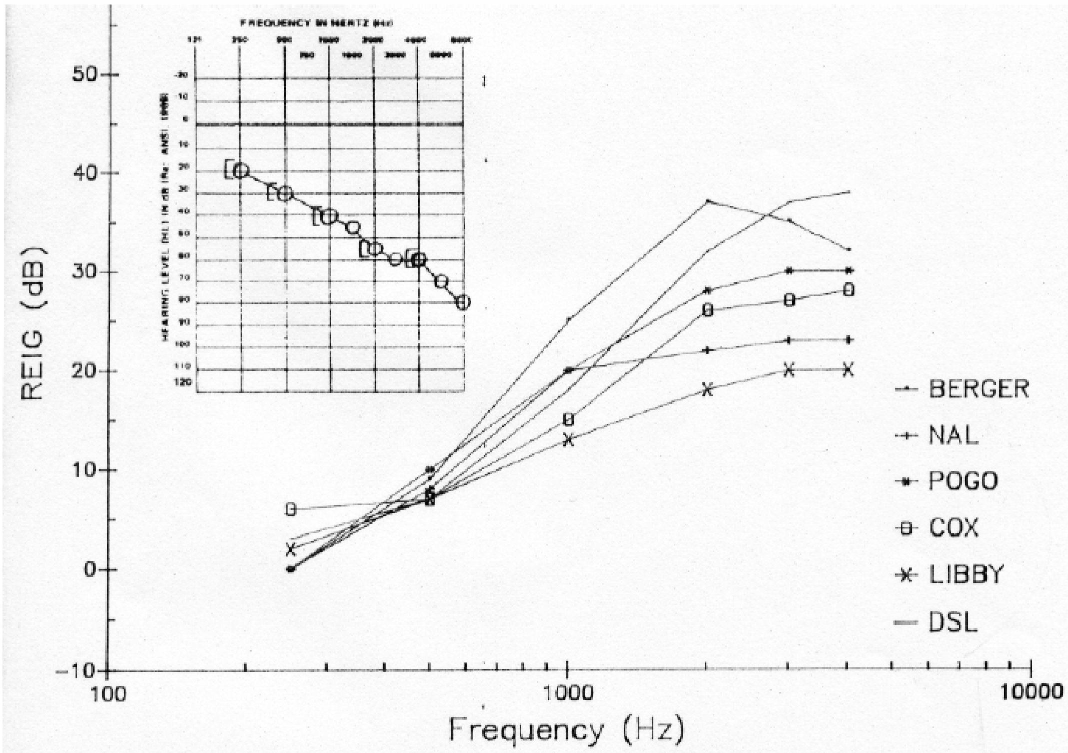

Figure 3.1. Targets for insertion gain, prescribed for a moderate sloping hearing loss by six different linear fitting rationales (Hawkins, 1992).

3.1.2 Non-linear gain prescription for varying input-levels.

During the late 1970's and in the 1980's, non-linear gain (or compression) began to be implemented in commercial hearing aids. Compression was initially used to prevent distortion at high output levels from the hearing aid. But gradually the compression threshold (CT) of the compressor became set to lower input levels, making it possible to compress larger parts of the dynamic input range (Barker & Dillon, 1999).

The general rationale behind automated gain regulation is to present sounds within the normal dynamic range, such that they become audible within the restricted range of the hearing-impaired listener. In a linear hearing aid, the gain prescription can be seen as only being valid for one given setting of the device's volume control (VC). That is, if the hearing aid is fitted to place speech spoken with normal vocal effort at the most comfortable level, the user may need to turn up the VC for soft sounds or turn it down for loud sounds. This is caused by the *undershoot* and *overshoot* of the linearly amplified loudness function, relative to normal loudness (shown in fig. 3.2, top).

With non-linear amplification, the gain is varied automatically such that soft sounds receive higher gain and loud sounds lower gain relative to the gain setting for medium-level inputs (shown in fig. 3.2, bottom).

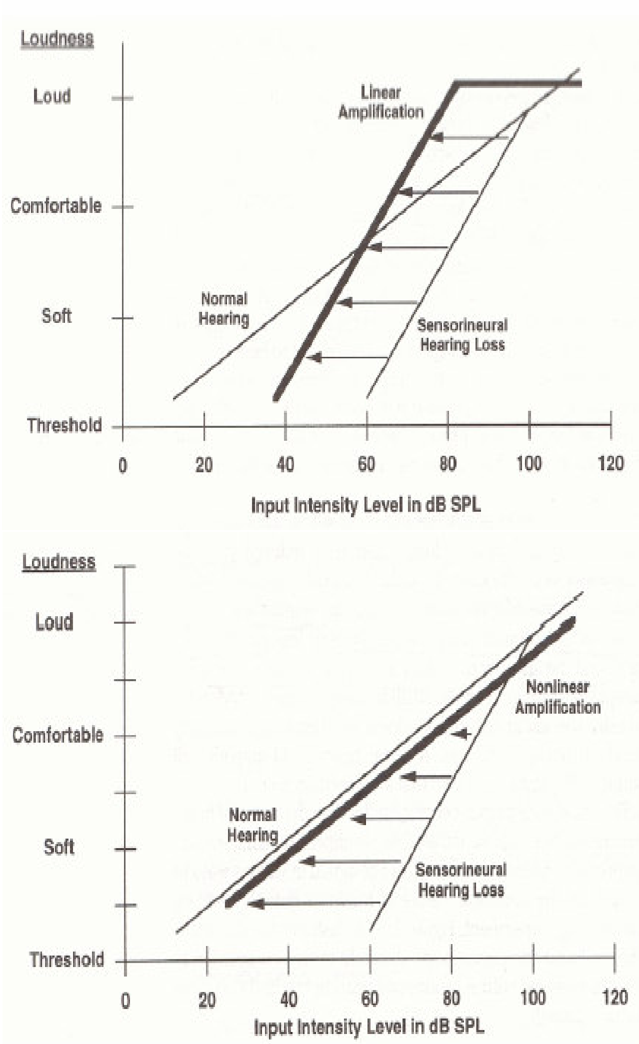

Figure 3.2. Illustration of the difference between linear and non-linear amplification (WDRC). In the top panel, the volume control has been set to provide comfortable loudness for an average speech input. In the bottom panel, gain is gradually reduced with increasing input-level in an attempt to match impaired loudness growth to the normal function (illustration from Stach, 1998).

At lower input levels, high amounts of gain are applied to make soft sounds audible to the listener. At higher input levels, gain is gradually diminished and finally reaches zero. At a given high input level, the output level from the device is approximately equal to the level of the direct sound path reaching the ear drum, through the ear mould and ventilation channel. In this way, the amplified loudness function will match the normal function, and (in theory) provide the listener with a natural loudness perception of low, medium and high-level sounds.

The main parameters used for characterising the dynamic properties of a compression system, are the *compression ratio* (CR), the *compression threshold* (CT), the *attack-time* (AT) and the *release-time* (RT). These parameters and their measures are defined in the IEC 118-0 standard (IEC, 1983) (see fig 3.3 and fig 3.4).

Depending on the fitting objective, the lower knee-point of the compressor may be set at different input-levels. In the case where the knee-point is set at a mid input level (e.g., 65 dB SPL), the compression-system is denoted as *Medium Level Compression* (MLC). When a very low compression knee-point is used (e.g., 20 dB SPL), and a wide range of input levels are compressed, the system is denoted as *Wide Dynamic Range Compression* (WDRC) (Dillon, 1996).

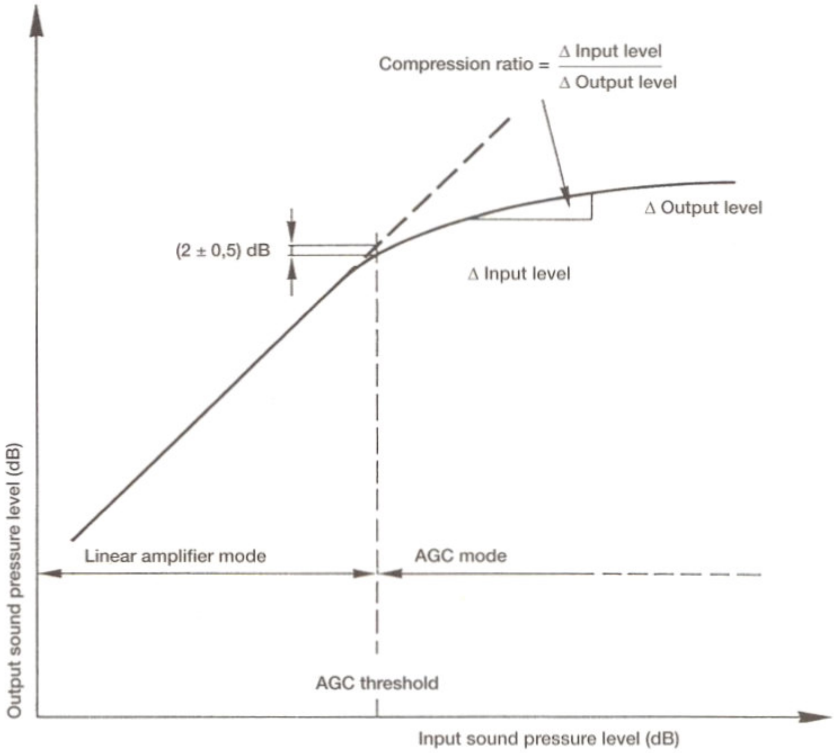

Figure 3.3. Illustration of the steady state input/output function measured with a pure-tone input at 1600 Hz. The *compression ratio* is the ratio of the difference between two input sound pressure levels and the corresponding difference in the output sound pressure levels (in dB). The *compression threshold* is the input sound pressure level at which there is a 2 dB reduction in the gain (+/- 0.5 dB) with respect to linear gain (IEC, 1983; illustration from Vonlanthen, 1995).

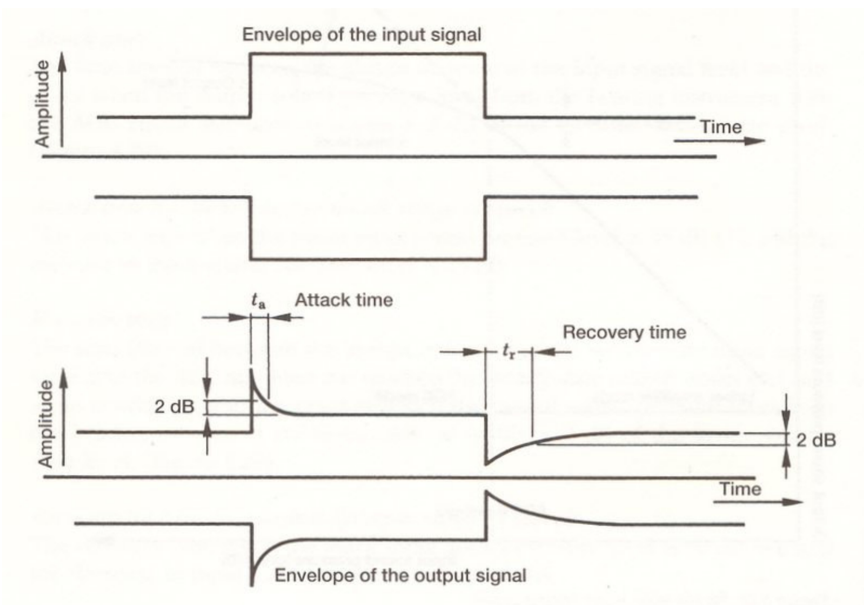

Figure 3.4. Illustration of the temporal aspects of the compressor. The attack-time is defined as the time in ms between an abrupt increase in the steady state input level and the point where the output level stabilizes within +/- 2 dB of the elevated steady-state level. Similar, the release time (or recovery time) is defined as the time interval in ms between an abrupt reduction in the input signal and the point at which the output level has stabilised again within +/- 2dB of the lower steady state level (IEC, 1983; illustration from Vonlanthen, 1995).

In figure 3.5, the two types are compared to linear amplification. Note that the "anchor-points" of all three systems are positioned at 65 dB SPL input level. That is, all three systems apply the same gain for a normal speech-input at a medium input-level - but the WDRC-system applies more gain for lower inputs and both the WDRC and MLC-systems provide less gain for higher input-levels, compared to linear amplification (as also shown in fig. 3.2).

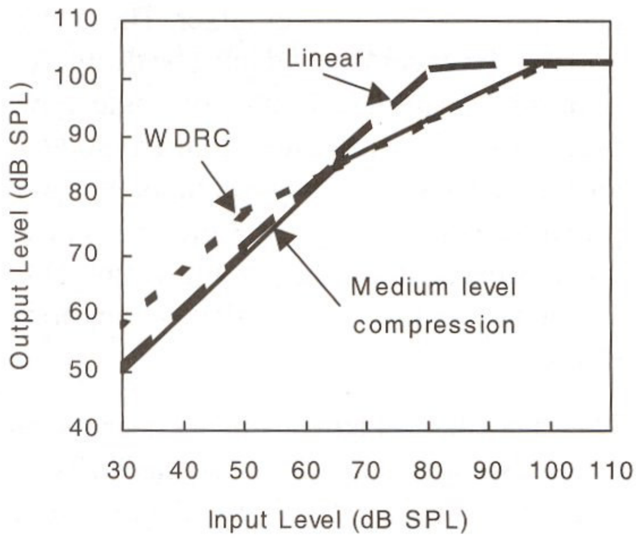

Figure 3.5. Steady-state input/output functions of linear gain, Wide Dynamic Range Compression and Medium Level Compression (illustration from Dillon, 2001).

Since the introduction of compression in hearing aids, several fitting rules for non-linear amplification have been developed. They include both generic rules, made to be applicable for all hearing aids on the market (e.g. FIG6 by Killion & Fikret-Pasa, 1993; DSL (I/O) by Cornelisse et al, 1995; IHAFF/Contour by Valente et al, 1997; NAL-NL1 by Byrne et al, 2001), and device-specific rules made by hearing aid manufactures as an integrated part of computer based fitting software (e.g. ScalAdapt by Kiessling et al, 1996; LPP by Phonak, 1999).

Some nonlinear fitting rationales are based on empirical or theoretical measures of loudness perception. Empirical measures include scaling of loudness on psychometric scales (e.g., Allen et al, 1990) and measurements of the most comfortable level and upper comfortable level as a function of hearing threshold (e.g., Pascoe, 1988). A theoretical measure of loudness has been obtained by modelling the transfer function of different cochlear stages and calculating the corresponding loudness in Sones for signals with different spectra (Moore et al, 1997).

The model by Moore and colleagues has been implemented as part of the NAL-NL1 rationale, which prescribes compression settings for non-linear hearing aids with up to four channels. This fitting rationale is further discussed in the following section, as it has been the scope for later investigations on preferred listening level for soft and loud sounds.

3.1.2.1 National Acoustic Laboratories Non-Linear Procedure

The aim of the NAL-NL1 procedure is to maximize speech intelligibility at a normal or less-than-normal overall loudness. A consequence of this aim is that some variation in the overall loudness is provided to the listener. But, in the absence of any data to indicate what loudness variation is desirable, NAL has adopted the principle of amplifying speech to normal loudness, or to a lower level if it provides greater speech intelligibility (Byrne et al, 2001).

NAL-NL1 was developed through analysing 52 different audiograms, resembling all common degrees and configurations of hearing loss. A computer model (depicted in figure 3.6) combined the effects of two components: the model for loudness perception by Moore et al (1997) and a modified version of the Speech Intelligibility Index.

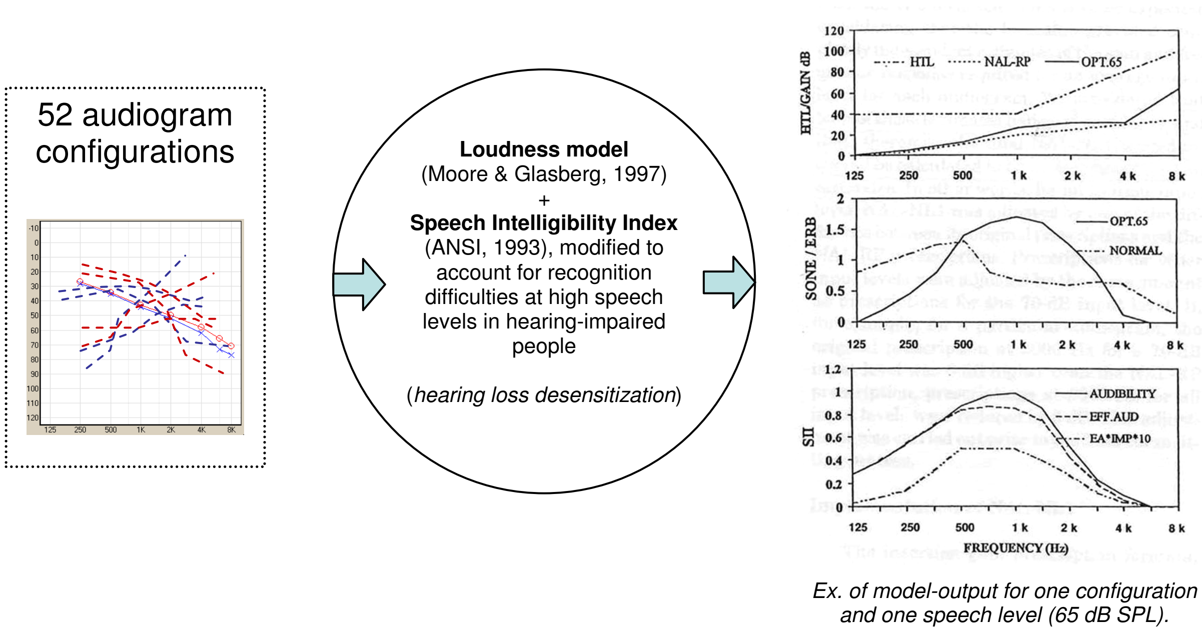

Figure 3.6. Schematic of development process of the NAL-NL1. Fifty-two audiograms were analysed by a MATLAB-model in regard to loudness and speech intelligibility. To the right, an example of the model output for one audiogram at one input level: The optimum frequency response, compared to the NAL-RP response and threshold (top), calculated loudness in Sones, compared to normal loudness (middle) and audibility calculated from the speech intelligibility index (bottom) (Byrne et al, 2001).

The modified index was based on research which showed that for people with severe losses, the SII overestimated speech recognition performance at high sensation levels (Ching et al, 1998). They also found that increasing the audibility in regions with excessive hearing loss, only lead to a further reduction in speech recognition. This reduction was believed to stem from other factors than audibility and the level distortion accounted for in the original SII standard (ANSI-S3.5, 1997) - possibly reduced frequency resolution and degraded temporal processing. Therefore, the original SII was modified with a hearing loss desensitization-factor in order to take these phenomena into account.

Based on the output from the computer model, a formula was developed that calculates optimal gain responses, based on the air and bone thresholds at each frequency, the "three frequency average" and the overall speech input level. The target screen from the NAL-NL1 fitting software is shown in figure 3.7. For a moderately sloping loss, NAL-NL1 specifies compression ratios from 1:1 to 2.5:1 and a compression threshold at 52 dB SPL. In the left panel, the resulting gain response for speech at 65, 80 and 90 dB SPL input-level is shown.

The response for the normal speech input is very close to the response prescribed by the linear NAL-R rule (Byrne & Dillon, 1986), which aimed at providing loudness equalisation to maximise speech intelligibility. It was not the intention with NAL-NL1 to achieve loudness equalisation, but the close resemblance of the two responses for an average input shows that it tends to do so between 500 Hz and 4 kHz, as a consequence of the maximising the SII (Byrne et al, 2001). In the right panel of figure 3.7, a simulation of the variation in the output spectra for a 65 and 80 dB SPL speech input is also shown.

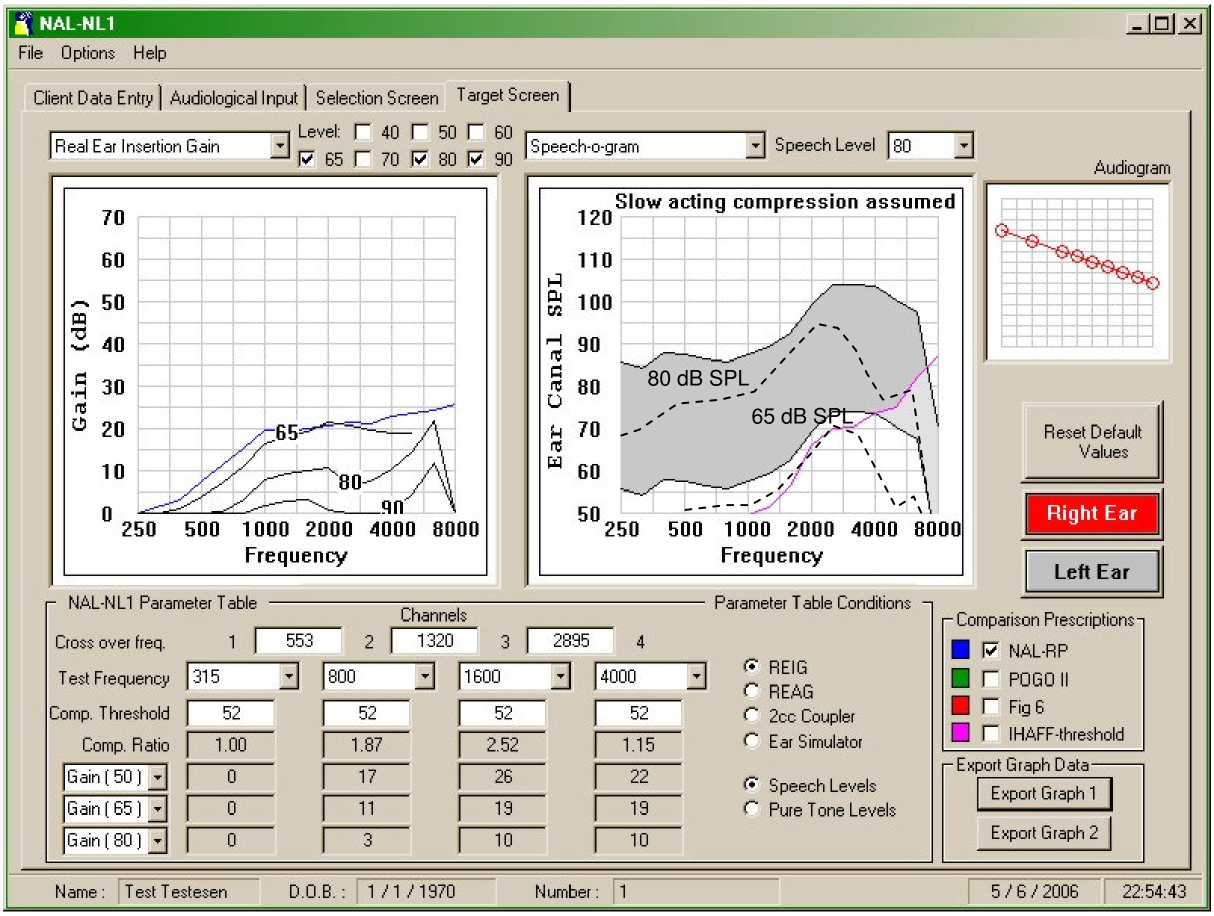

Figure 3.7. Target screen from NAL-NL1 fitting software. In the left panel, insertion gains for three speech input-levels of 65, 80 and 90 dB are shown. The right panel shows the variation in the output spectra for a 65 and 80 dB SPL speech input (Screen capture from NAL-NL1 fitting software)

3.2 The effect of compression parameters on the sound processing

Many generic fitting rationales, like NAL-NL1, do not recommend specific time constants in their prescriptions. This is still an issue of debate. Byrne et al (2001) argue that no clear evidence exists whether fast or slow time constants is best for providing better intelligibility and listening comfort.

Indeed, several factors influence the way a signal is processed by the compressor. The interaction between the compression ratio, time constants and the type of input signal should be touched upon here. When the input signal to the compressor is a modulated speech signal, the steady state input-output characteristics at a given frequency can only be obtained with very short attack and release times (e.g., 1-5 ms). In that case, the AGC will be able to follow (and compress) the fast modulations in the signal, and thus the effective compression ratio is relatively high (Verschuure et al. 1996).

The dynamic input/output-function in such a system will partly coincide with the steady-state function, as shown in figure 3.8.

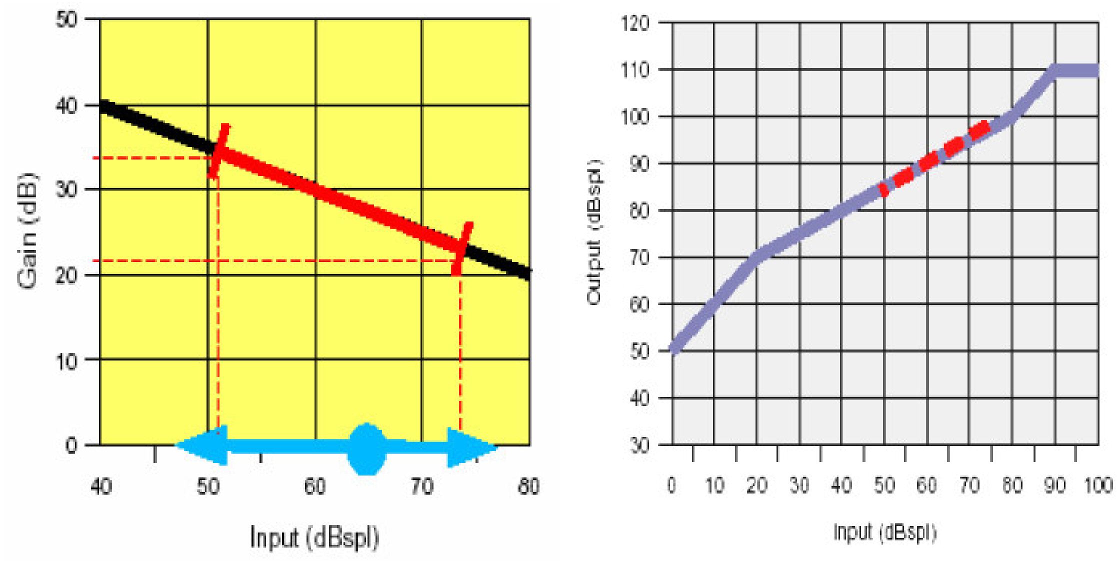

Figure 3.8. Input-gain graph (left) and input-output function for a hearing aid with very short attack and release times. The arrow shows the dynamic range of speech with the dot being the RMS-level. The gain variation, when the attack and release times are short (e.g., 1-5 ms), is indicated on the input-gain curve. To the right, the dynamic input-output function of this system (dashed line) relative to the static function (own illustration).

When longer time constants are used, the AGC will only be able to follow slower modulations in the signal. Depending on the combination, either being (1) short attack and long release times, (2) equally long attack and release times or (3) long attack and short release times (although less common), the gain will stabilise according to the overall level of the input signal.

In these three cases, the effective compression ratio is relatively low, and the dynamic input/output-functions approach linear gain, as shown in figure 3.9. With the long release time in example (1), the overall output level from the hearing aid will be lower with a speech input, compared to a steady-state input signal.

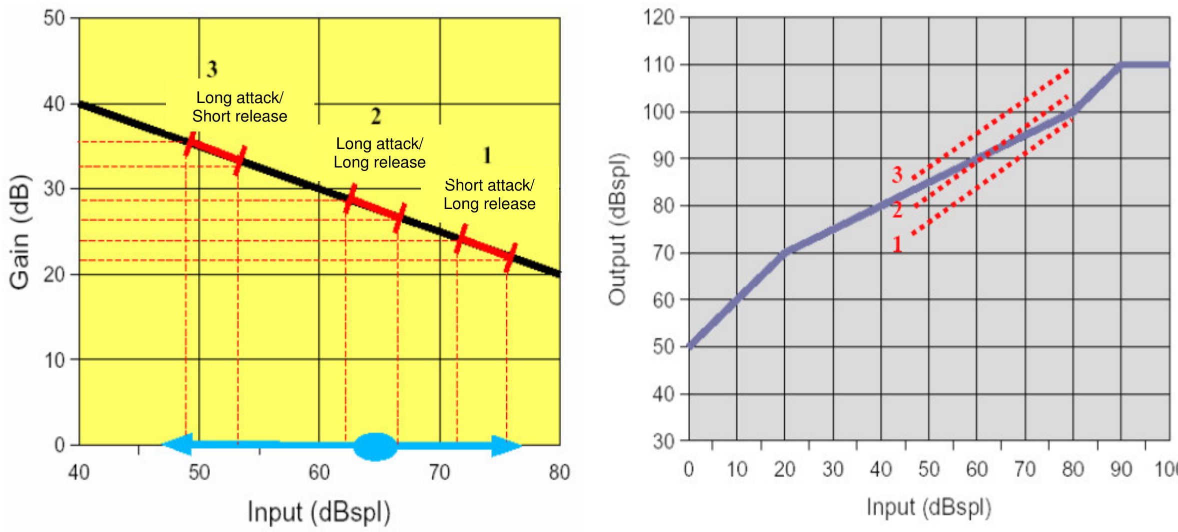

Figure 3.9. Input/gain graph (left) and input/output function (right) for a hearing aid with three combinations of short and long release time constants. The arrow shows the dynamic range of speech with the dot being the RMS-level. The gain variation in the three situations is indicated on the input/gain curve. To the right, the corresponding dynamic input-output functions (dashed lines) relative to the static function (own illustration).

When the input signal is speech alone or speech in environmental noise, the relationship between compression ratio and time constants (along with other parameters) may affect the intelligibility of speech, as well as the perceived sound quality and listening comfort. This issue also has relevance for the processing of loud sounds. In the following section, some earlier studies investigating the perceptual effects of compression will be reviewed, as parts of the methodology in these studies were used for the experiments in this project.

3.3 The influence of compression on speech intelligibility, attributes of sound quality and listening comfort

One may isolate one or two of the compression parameters to assess its perceptual influence on the output signal, while keeping other parameters constant in the experimental situation. This has been done for speech and noise inputs to single and multichannel compressors.

3.3.1 Investigations by Neuman et al, using a single band compressor.

Neuman et al. (1994), asked hearing-impaired subjects to judge the sound quality in a paired comparison test. They presented speech in different levels of background noises, through a "slow acting" single band compressor. The attack time (AT) was 5 ms and release time (RT) 200 ms, while the compression ratio (CR) varied from 1:1 to 10:1. The investigators found that subjects had a significant preference for compression ratios below 3:1. For the high noise levels, subjects preferred a low CR of 1.5:1 or linear gain, compared to the lower noise levels where they preferred a CR up to 2:1.

In a new study, Neuman et al (1995a) looked at the effect of varying the release-time in combination with different compression ratios. They processed speech in noise at various positive signal-to-noise ratios, in a single band compressor. Release times spanned from 60 ms to 1000 ms and three CR's of 1.5:1, 2:1 and 3:1 were used. No significant main effect for the release time was found, but there was a significant interaction between release time and noise level. Subjects preferred longer release-times as the noise level increased.

The preferences seen in these two studies may be explained by temporal changes in the noise-level. With a high CR (and short RT), the background noise will be amplified more in the speech pauses (denoted as the *pumping effect*), making the signal annoying to the listener and possible reducing speech intelligibility. On the other hand, with a low CR (or a longer release-time), the level difference between speech and noise segments in the same signal will become greater. Waveforms of a speech and noise signal, processed with a short versus a long release time, are shown in figure 3.10.

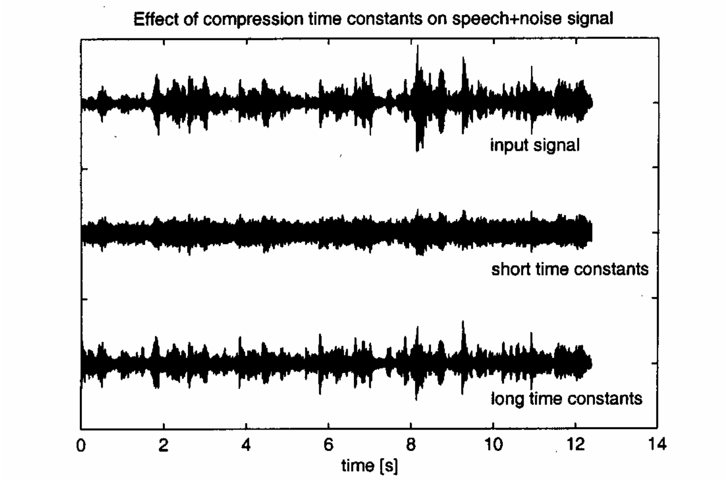

Figure 3.10. Waveforms of a speech and noise signal, processed with a short versus a long release time. The original input-signal is shown in the top-panel (Hansen, 2002).

In a later study, Neuman et al. (1998) looked further into the perception of sound quality, when the compression ratio and time constants were varied in a single channel compressor. They asked 20 hearing aid users to rate the *clarity*, *pleasantness*, *background noise*, *loudness* and *overall impression* of the compressed signals. Subjects rated their impressions on categorical scales with 10 major marks, as used by Gabrielsson et al (1990).

Test signals consisted of speech presented at 20 dB (RMS-level) above the compression threshold, and three types of background noise presented at -15 dB (ventilation), -5dB (apartment) and +5 dB (cafeteria) relative to the threshold. After compression, all signals were matched in regard to loudness by equating the 90th percentile of the cumulative distribution of the compressed speech to the same point on the cumulative distribution of the uncompressed speech signal (Levitt & Neuman 1991, Bakke et al. 1991). Finally, test signals were amplified to NAL-R targets for the speech (Byrne & Dillon, 1986) and presented over headphones. The aim of this procedure was to present all processed signals, such that speech was perceived as having a comfortable listening level.

Two experiments were conducted. In the first experiment the AT of 5 ms and RT of 200 ms were kept constant, and the CR was varied from 1, 1.5, 2, 3, 5 and 10:1. In the second experiment, the AT of 5 ms was held constant while the RT was varied (60, 200 and 1000 ms) and the CR was varied (1.5, 2 and 3:1).

The two experiments showed that varying the compression ratio had the greatest impact on the ratings. The rating of *clearness*, *pleasantness* and *overall impression* dropped, and the rating of *background noise* increased significantly, as a function of increasing ratio. Overall good sound quality was preserved when the ratio was below 3:1. The effect of varying the release-time was more subtle. There was no significant effect of release time for CR ≤ 2:1. But for CR = 3:1 a short release time of 60 ms gave a significantly lower rating of *clearness*, *pleasantness* and *overall impression* and an increase in the *background noise*, compared to the 200 ms and 1000 ms conditions. This was especially the case with the cafeteria noise. Similar results were found in Neuman et al. (1994, 1995a).

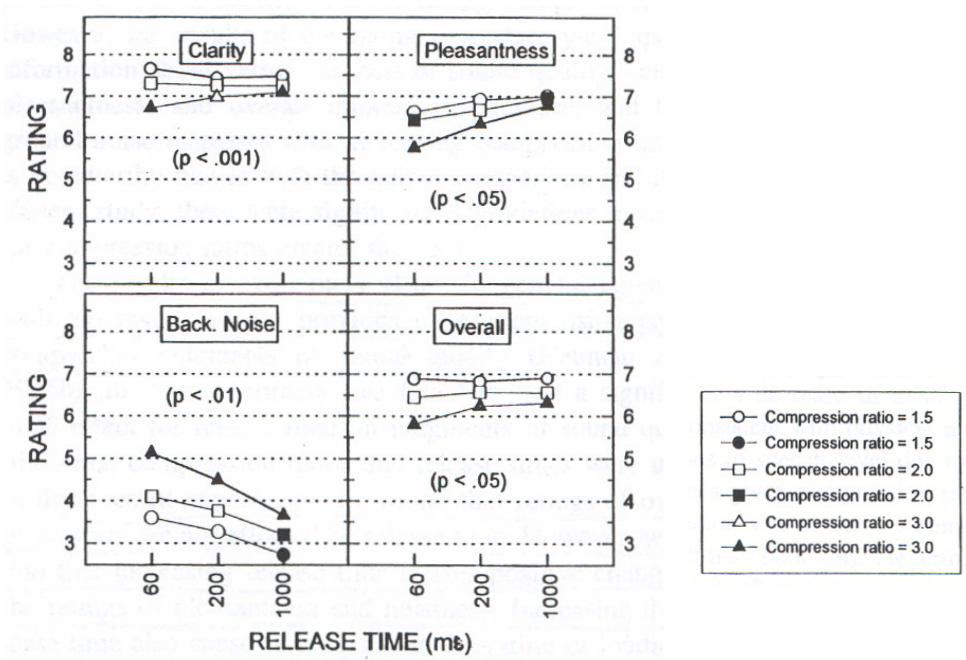

Figure 3.11. Mean ratings of clarity, pleasantness, background noise and overall impression, as a function of compression ratio and release time. Mean scores within a compression ratio found to differ significantly ($p < 0.05$) are indicated by filled symbols (Neuman et al, 1998).

In all three studies by Neuman and colleagues, a single channel compressor was used. Most commercial hearing aids on the market today use several compression channels, whose filter skirts are partly overlapping over the frequency range. This approach is based on models of the division of peripheral cochlear processing into critical bands. By dividing the hearing aid's sound processing into bands of approximate critical bandwidth, it should in theory be possible to restore parts of the non-linear processes in damaged cochlear filters (Moore et al, 1999). Also, the frequency response may be more precisely adjusted in accordance with the shape of the individual audiogram.

3.3.2. Investigation by Hansen, using a multi-channel compressor.

Depending on the compression settings, a multichannel set up may affect the signal differently compared to a single band compressor. In a single band compressor, the gain-variations are controlled by the more powerful parts of the signal, often present at the lower frequencies. With many channels, the compression in each channel will only depend on the input to that channel.

Hansen (2002) investigated the effect of attack- and release-times on the subjective impressions of sound quality and speech intelligibility. He processed real-life speech- and noise-signals, recorded binaurally at ear level, through a simulated hearing aid with 15 independent compression channels.

The gains of the individual compressors were adjusted based on the measured hearing threshold values. This was accomplished by calculating the gain prescribed for the hearing loss using the NAL-R rule. The compressors were then adjusted so that this prescribed insertion gain would be reached for an (overall) input level that equals the standard long-term average level of speech with a normal effort in the respective band. This means that the input level to the individual bands would be less than the overall speech level, because the summed inputs of all bands together should be equal to the overall input level of 62.4 dB SPL (i.e., the level for speech at normal vocal effort, according to ANSI-S3.5, 1997). After compression, all signals were matched to the same RMS-level, in order to avoid level-differences between signals to act as listening cues.

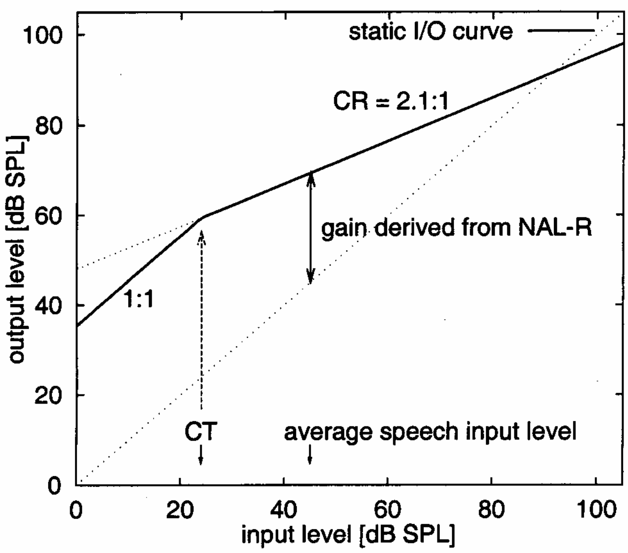

Figure 3.12. Static input/output-function for each compressor in the simulated hearing aid. The broadband input level of normal speech is assumed to be 62.4 dB SPL. The input level in each channel (shown by the small arrow) is lower than the overall level, due to the splitting of the intensity into many channels (Hansen, 2002).

The test group comprised of six hearing-impaired persons with moderately sloping losses and six normal-hearing persons. They rated signals in a paired-comparison test, indicating the degree of preference for either signal A or B on a computer screen. The rating was done on a seven-point scale, with the midpoint indicating no preference for either signal A or signal B.

In a first experiment, four hearing aids with varying attack times (1, 10 and 100 ms) and release-times (40, 400 and 4000 ms) were compared. The compression ratio was fixed at 2:1 (see fig. 3.13, left panel). All subjects showed a significant preference for the longest release-time of 4000 ms, both in regard to sound quality and speech intelligibility. There was no significant difference in ratings when the attack-time was changed from 1 to 100 ms (HA#3 and 4). Apart from real-life speech and noise signals, some music signals were also used as test signals. There was a greater standard deviation in the ratings for the music-signals, and subjects also indicated that they had problems telling the difference in sound quality and intelligibility of these signals.

In a second experiment the attack-time was held constant, while the release-time, compression threshold and ratio were varied. The AT was 1 ms in all conditions, while the RT varied between 40 and 4000 ms, the CT between 20 and 50 dB SPL and the CR between 2.1:1 and 3:1 (fig. 3.13, right panel).

TABLE 1. Values of attack times (AT) and release times (RT) for the four hearing aid settings.

HA #	1	2	3	4
AT [msec]	1	10	1	100
RT [msec]	40	400	4000	4000

Within one setting, the values for AT and RT were applied to all compressor channels

TABLE 2. Values of attack times (AT), release times (RT), compression threshold (CT), and compression ratio (CR) for the four hearing aid settings.

HA #	1	2	3	4
AT [msec]	1	1	1	1
RT [msec]	40	40	40	4000
CT [dB SPL]	20	50	50	20
CR	2.1	3.0	2.1	2.1

Within each setting, the values for AT, RT, CT, and CR were applied to all compressor channels

Figure 3.13. Compression settings used in experiment 1 (left) and experiment 2 (right), in the study by Hansen (2002).

In regard to sound quality, the hearing-impaired subjects gave the highest rating to HA# 4 with the longest RT and lowest CT. HA# 2 and 3 received the lowest ratings, partly because the high CT made the soft speech inaudible for this group. The normal-hearing subjects showed no significant difference in regard to sound quality, but the trend of HA# 4 receiving the highest rating was also seen here. The ratings of perceived speech intelligibility showed the same trend. Here both groups rated HA#4 significantly higher, than the three other hearing aids.

In summary, the study by Hansen (2002) showed that both hearing-impaired and normal-hearing subjects preferred a longer release-time in combination with a low compression threshold and a compression ratio of 2:1. This is partly in agreement with Neuman et al (1995a, 1998), who also observed this for some noise types using a single channel compressor.

An interesting aspect of the Hansen-study is that he used several types of real-life speech and noise signals, as well as music. The speech-recordings comprised of conversations at a train station, in a cafeteria, at a workplace and pre-recorded speech (Dantale I) mixed with a real-noise signal. The input level in the recordings varied depending on the situation and the distance of the HA-microphone to the sound source. Still, with these very different signals, the data analysis revealed some very significant trends from the experiments. Also, the use of real life signals containing varying sound levels is preferable for creating realistic listening situations, compared to test signals with steady presentation levels for speech and noise.

Continuing along this path, it seems relevant to investigate the perceptual effects of compression, with real life signals at varying sound pressure levels. It should be investigated how non-linear gain prescription for soft and loud sounds affect the perception of e.g., speech intelligibility, sound quality and listening comfort. The question is what amount of gain should be applied for varying input levels, and whether the concept of *loudness normalisation* is in fact a right approach for mapping the normal dynamic range into that of the hearing-impaired listener.

3.4 Earlier investigations on the preferred listening levels for soft and loud input signals

Although many authors note the lack of knowledge on how to process soft and loud input-levels (e.g., Neuman et al, 1998; Byrne et al 2001), very few studies have dealt with the user preference of gain prescriptions for varying input levels. In the following, two studies investigating loudness and the *preferred listening level* (PLL) for soft, medium and loud input levels are described.

3.4.1 Investigation by Neuman et al, regarding preferred listening levels.

Neuman et al (1995b) investigated the relationship between the most comfortable listening level and preferred listening levels for speech and noise at various presentation levels. They presented continuous speech at three levels (55, 70 and 85 dB SPL) in three different background noises (ventilation at 50 dB SPL, apartment at 59 dB SPL and cafeteria at 71 dB SPL) - yielding nine combination of signal-noise-ratios, over the 30 dB speech range.

Signals were processed in a single-channel compressor, with six different compression ratios (1:1, 1.5:1, 2:1, 3:1, 5:1 and 10:1). The compression threshold was 65 dB SPL peak-level. The attack time was 5 ms and the release time 200 ms. For each compression condition, gain was adjusted to place speech of 70 dB SPL at the most comfortable level of the listener. This was a direct implementation of the NAL-R fitting rationale (Byrne & Dillon, 1986). An example of input/output-functions adjusted to the hearing loss of one subject is shown in figure 3.14.

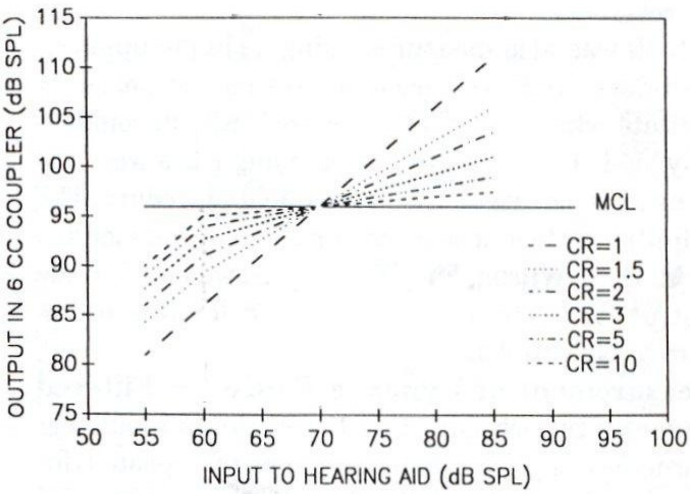

Figure 3.14. Input/output functions for a single subject, measured in a 6 cc-coupler. The gain for all compression conditions were adjusted to NAL-R for a speech input of 70 dB SPL. RMS-level (Neuman et al, 1995b).

Signals were presented to 20 hearing-impaired subjects over headphones. First, the most comfortable listening level for speech, as perceived through the linear hearing aid (1:1) was measured in each individual. Secondly, all subjects were asked to listen to the individual signals, and indicate if they would want to adjust the volume control in a real listening situation. In case of a yes, subjects were then asked to adjust the signal to the preferred level for satisfactory listening. The average deviation from MCL (in dB) for the three speech levels and three noise types could then be calculated. The interactions between mean PLL's for speech level and compression ratio and for noise type and speech level are shown in figure 3.15.

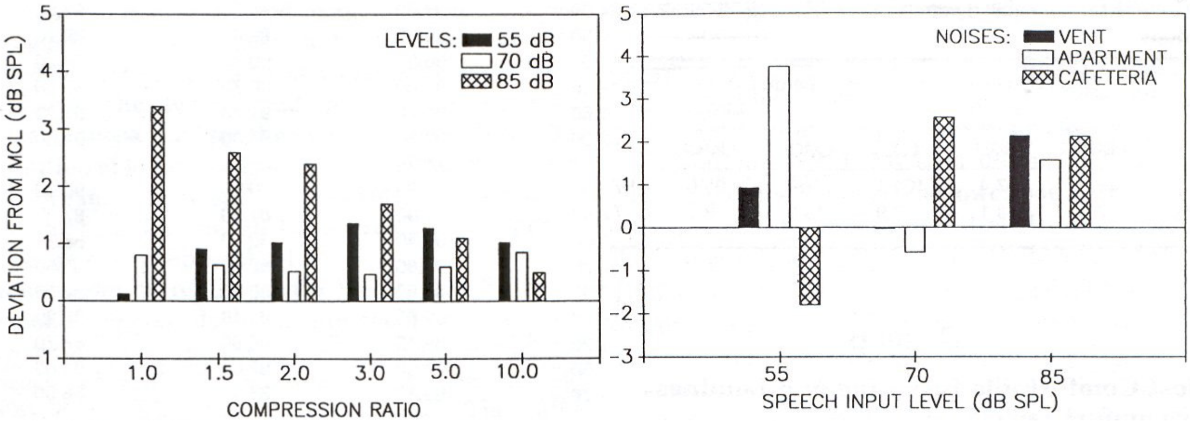

Figure 3.15. Left: Mean deviations from MCL for the three speech levels as a function of compression ratio. Right: Mean deviations from MCL for the three noise types is shown at each speech level (Neuman et al, 1995b).

In all conditions, the mean PLL's were found to be no more than +/- 5 dB relative to the most comfortable level. A repeated measures analysis showed that the dynamic range of the listener, the noise type and the input level all had small but significant influence on the deviation from MCL. Subjects with small dynamic ranges preferred listening levels slightly below the MCL, whereas subjects with large ranges preferred levels slightly above MCL. Signals with high input levels and better signal-to-noise ratios were adjusted to higher listening levels, compared to signals with poorer signal-to-noise ratio that were adjusted closer to the MCL.

The authors suggest that, depending on the speech level and signal-to-noise ratio, a slow acting compression hearing aid should place the output within 5 dB of the MCL. This should be done with the combination of a mild compression ratio (to avoid degrading speech quality) and gain than places speech at 70 dB SPL RMS-level at the user's most comfortable level.

3.4.2 Investigations by Smeds et al, regarding preferred loudness

The National Acoustic Laboratories Non-linear procedure (Byrne et al, 2001) has also been subject to investigation of preferred listening levels. In a series of experiments, Smeds et al (2004a) looked upon loudness aspects of prescriptive methods for nonlinear hearing aids. In one laboratory experiment, eleven speech and environmental sounds at various presentation levels were processed according to the NAL-NL1 procedure (see subsection. 3.1.2.1).

Compression parameters were prescribed using stand alone software (see fig. 3.7). The NAL-NL1 procedure does not prescribe settings of time constants, but Smeds et al used attack and release times of 2000 ms. This was done to simulate a slow acting compression system that changes its gain when the listening situation changes, but does not change the gain within individual situations.

dB SPL	Non-speech situations	Speech situations
46	Bush walk	
54		Quiet conversation, baby asleep
58	Fan noise, office worker	
61		Conversation indoors
65		Conversation outdoors
70	Small gathering, babble	
71		Conversation in babble noise
75		Speech in vacuum-cleaner noise
81	Motorway, outdoors	Arguing
86	Sawing	

Table 3.1. Listening situations used in Smeds et al (2004a)'s evaluation of NAL-NL1.

The processed signals were presented in a sound-isolated room. 15 normal-hearing and 24 hearing-impaired listeners participated in the study. Test subjects were seated in front of a TV-set and two loudspeakers. Test signals and video had been recorded in actual settings and comprised of eleven different situations (for example a bush walk, normal conversation, speech in vacuum-cleaner noise, motorway etc.). The presentation levels ranged from 46 – 86 dB(C).

First, the signals were presented one at a time, and subjects were asked to rate the loudness and indicate their interest in the signal. Secondly, subjects were then asked to adjust the presentation level to the preferred listening level. Finally, loudness rating and indication of interest were performed again, using the adjusted presentation level.

Results showed that both normal and hearing-impaired listeners preferred less than normal loudness (which was calculated by the model by Moore et al, 1997). This was especially the case for the loud signals (60 – 79 dB(C)). For the hearing-impaired group, the loudness level for speech situations within 60-79 dB(C) were reduced by 11-14 phon relative to normal loudness. The difference in Phones between preferred and normal loudness for the eleven situations are shown in figure 3.16.

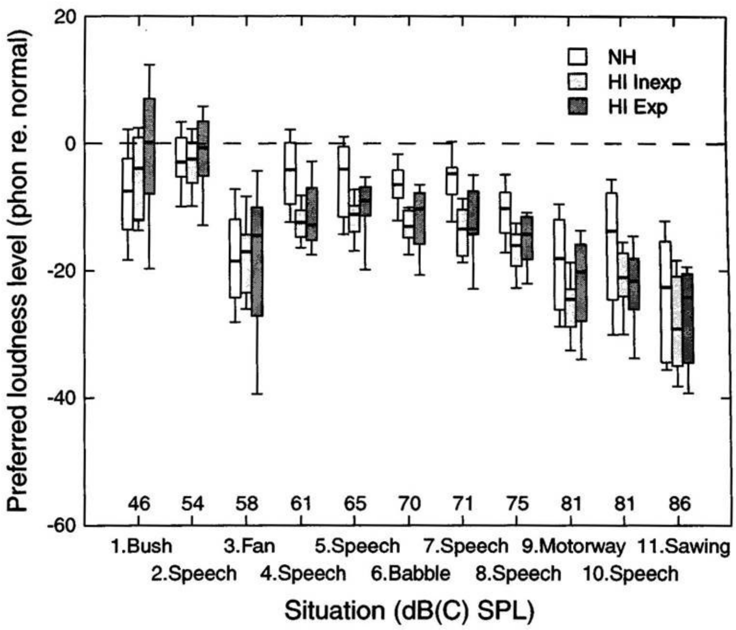

Figure 3.16. Deviations from normal loudness (calculated with the model by Moore et al, 1997) shown for the eleven listening situations. Medians, inter-quartiles, maximum and minimum values are shown for the three groups; Normal-hearing persons, experienced HA-users and inexperienced HA-users (Smeds et al, 2004a).

Loudness scaling of individual signals also showed that the hearing-impaired group adjusted presentation levels such, that subsequent loudness ratings for soft and loud signals became clustered around the moderate loudness category (figure 3.17). Especially, loudness ratings for the high-level signals were substantially lower, compared to ratings made before the level adjustments. In the normal-hearing group, subsequent loudness ratings were only reduced for signals with highest presentation levels.

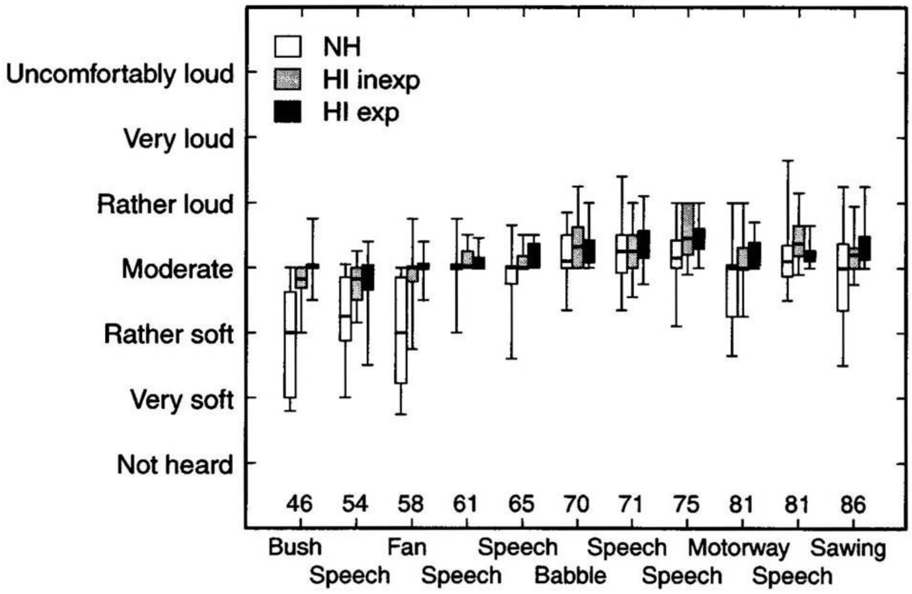

Figure 3.17. Ratings of loudness, after adjusting volume to the preferred listening level. Medians, inter-quartiles, maximum and minimum values are shown for the three groups; Normal-hearing persons, experienced HA-users and inexperienced HA-users (Smeds et al, 2004a).

In a following field trial (Smeds et al, 2004b), all subjects were fitted with research hearing aids, according to NAL-NL1. Subjects wore the hearing aids for one week, and were asked to adjust the volume-control to their preferred listening level. All adjustments were logged by the hearing aid. The results showed that on average the normal-hearing subjects did not adjust the volume control, whereas the hearing-impaired preferred gain reduction in most cases, leading to less loudness than prescribed by the NAL-rationale.

3.5 Summary and suggestions for listening experiments investigating hearing aid processing of loud sounds.

In summary, investigations of the effects of compression settings show a preference for longer release times. When shorter release times are used, the sound quality and speech intelligibility can be maintained if the compression ratio is below 3:1 (subsections 3.3.1 and 3.3.2).

When focusing on preferred loudness for soft and loud sounds, the studies by Neuman et al (1995b) and Smeds et al (2004a, 2004b) show that hearing-impaired listeners may prefer less loudness than would be expected by a commonly used loudness normalisation rationale. This points towards a problem with the loudness model used to derive the fitting formula - that is, the model seems to underestimate overall loudness for a given input-signal, especially for high input levels. The objective for NAL-NL1 is to reduce loudness to lower than normal levels, if this is beneficial for the intelligibility of speech (as calculated from the SII). Thus, for some of the loud signals in the study, the loudness was already lower than normal, before level adjustments took place, and still listeners prefer it to be even lower.

One interesting thing about the study by Smeds et al. was that no significant difference in preferred listening levels was seen between experienced and inexperienced hearing users. This is partly in contradiction with other studies, showing that experienced users are capable of tolerating higher gain settings during daily hearing aid use (e.g., Marriage et al, 2004).

Smeds et al. recommend that care should be taken not to prescribe too much gain in general, for mild to moderate hearing losses. Similar conclusions are made by Neuman et al (1995b), who states that presentation levels at the ear drum should be within +/- 5 dB of the most comfortable listening level. Thus, when focusing on loudness and the preferred listening level for loud signals, there may be a preference for a lower loudness than suggested by commonly used fitting rationales - possibly requiring a high compression ratio in combination with longer time constants (i.e., a less effective compressor), that places the signals close to the listener's most comfortable range. On the other hand, when focusing on the quality and intelligibility of speech, there seems to be a preference for compression ratios no higher than 3:1, possibly combined with a long release time of the compressor.

Based on the reviews in this chapter, it is clear that the conclusion on what compression settings are best, might depend on the question asked – that is, whether focus is on preferred listening level and loudness, or on the subjective impressions of sound quality and speech intelligibility. In addition, the type of compression system used in the investigation will also influence the results and conclusions made.

In any case, it seems apparent that some variation in the output level should be provided, in order to let the hearing aid user experience a difference between various speech levels, as well as the levels of different environmental sounds. But the variation in output level needed to obtain this effect may be much less than what is expected to restore "normal loudness". It may be that restoring normal loudness, as calculated by a model, is not the right goal. Rather the hearing aid rationale may only need to "indicate" the level difference - the amount of level-variation being dependent on the hearing loss and the dynamic range available in a given user.

Within the scope of this project, it should be investigated how non-linear gain prescription for loud sounds affects the perception of e.g., speech intelligibility, sound quality and listening comfort. A natural order of such an investigation would be, first to study the preferred or acceptable listening level for loud sounds. Secondly, the influence of the dynamic aspects of compression should be investigated. And thirdly, the influence of the shape of the frequency response should be investigated, e.g., to study whether listeners prefer more or less high frequency gain for loud input-signals.

In the following two chapters, experiments investigating the first two of the aspects mentioned above are described «««.

4. Perception of level variations in loud speech and noise signals, processed by a simulated non-linear hearing aid (experiment #1)

4.1 Introduction and research questions

The objective of the present study is to investigate hearing-impaired listeners' perception of level variations in loud speech and noise signals. Common non-linear fitting rationales seek to compensate for loudness recruitment by normalizing the perceived loudness for soft, medium and loud sounds in all frequency bands. This implies that input sounds may be presented in all parts of impaired listener's dynamic range - from the threshold to the upper comfortable loudness levels.

Previous research has suggested that hearing-impaired listeners prefer listening levels for soft and loud sounds to be closer to the most comfortable loudness level, than would be suggested by a typical loudness normalisation-schemes (Neuman et al, 1995b; Smeds, 2004a, 2004b). This would mean that care should be taken, at least when prescribing gain for sound levels beyond the level yielding a comfortable loudness sensation in impaired listeners.

The issue of preferred gain settings for high-level sounds may be investigated in both field trials and laboratory experiments. Laboratory experiments can help in the development of clinical tools useful for the validation of hearing aid fittings, regarding the processing of soft and loud sounds. When investigated in the laboratory, this topic poses a challenge in regard to applying a relevant method for investigating the perception of loud signals amplified by the hearing aid.

In the studies by Smeds and Neuman et al, subjects were asked to adjust the volume control to the *preferred listening level*, when listening to continuous speech and environmental sounds at fixed presentation levels. One may question whether listening to continuous loud signals resembles a realistic listening situation. In real life, presentation levels could also be fluctuating over time depending on the listening situation, the sound source and the distance relative to the listener. This might change the listener's tolerance for loud signals, compared to a situation where signals are presented at a high, fixed level.

The present study proposes an alternative approach, where test signals with built-in level variation are used to investigate the perceptual effect of hearing aid processing of loud speech and noise signals. Specifically, this method was used to investigate the relationship between compression ratio and listeners' impression of the level variation, loudness and their acceptance of the processed loud signals.

Also, the study investigated whether spectral differences among signals having equal overall RMS-levels, influence listeners acceptance of the level variation, when signals are processed with the same gain and compression-settings. If that is the case, this might imply the need for a fitting rationale that takes into account the input spectrum when prescribing real-time gain targets for the hearing aid. Such a fitting rationale would be in accordance with an objective of always keeping the degree of user-acceptance (or listening comfort) as high as possible.

In addition, the present experiment could also provide insight into the influence of signal duration on the tolerance for loud sounds. Due to differences in test method, the loud signals in this study have a shorter duration than signals used in the previous studies. Although not entirely comparable, the results of this study may be indirectly compared to the ones obtained by

Neuman et al (1995) and Smeds (2004a, 2004b). They used loud signals at similar presentation levels, but in their test method, signals were presented for longer time periods, while subjects assessed their loudness. This might influence listeners' preferences and tolerances for loud signals, yielding a lower preferred listening level compared to a situation where loud signals are presented for shorter periods of time.

To clarify, the present study attempts to answer the following research questions:

- What is the relationship between compression ratio and hearing-impaired listeners' impressions of level variation, loudness and acceptance, when loud signals are processed by a simulated non-linear hearing aid?
- Do spectral differences among signals with equal overall RMS-levels, influence hearing-impaired listeners' acceptance for loud sounds, when signals are processed by the same hearing aid?
- Does signal duration along with differences in test methods, influence the listeners' acceptance of loud signals?

4.2 Method

4.2.1 Input signals

Four different input signals were prepared in a sound-editing program (Adobe Audition, version 1.0). Three of the signals contained a mix of speech and noise, and a fourth signal contained a purely environmental signal.

(1) Dantale speech at normal vocal effort & party-noise (+ 5 dB SNR).
(2) Dantale speech at normal vocal effort & car cabin-noise (+ 5 dB SNR).
(3) Female and male speakers at normal, raised and loud vocal efforts & party-noise (+ 5 dB SNR).
(4) Audience noise from a football match.

Each signal had a duration of 30 seconds and contained four segments, each with a duration of approximately 6 seconds. The RMS-levels of the four segments were adjusted such that there was a 20 dB level variation in the overall signal[1]. That is, the RMS-levels of segments (2) and (3) were 10 dB and 20 dB respectively above the RMS-level of segment (1). The RMS-level of segment (4) was 10 dB lower than the level of segment (3). In this way the total signal had a rising and falling level contour, which was appropriate for the repetitive presentation of signals during the listening test. In the three signals containing speech and noise, the signal-to-noise ratio in each of the four segments was kept at +5 dB.

Sound examples of the four input signals can be found on the audio-CD in appendix 9.1. A schematic illustration of the format used for the signals is shown in figure 4.1. In the following section a more detailed description of the four signals is provided.

[1] The root mean square levels for the input and test signals were measured in the sound editing software used for generating the test signals (Adobe Audition, 2003). Measurements were done by selecting all parts of the signal (apart from the initial fade in and final fade out) using a window-width of 50 ms. From this selection the software calculates three RMS-levels, the minimum, the maximum and the average level. The minimum and maximum RMS-levels are the lowest and highest window-values found in the chosen selection. The average RMS-level is the average of all of the sums of the minimum and maximum values from the window sections in the selection (S. Garnett, personal communication, May 17th, 2006).

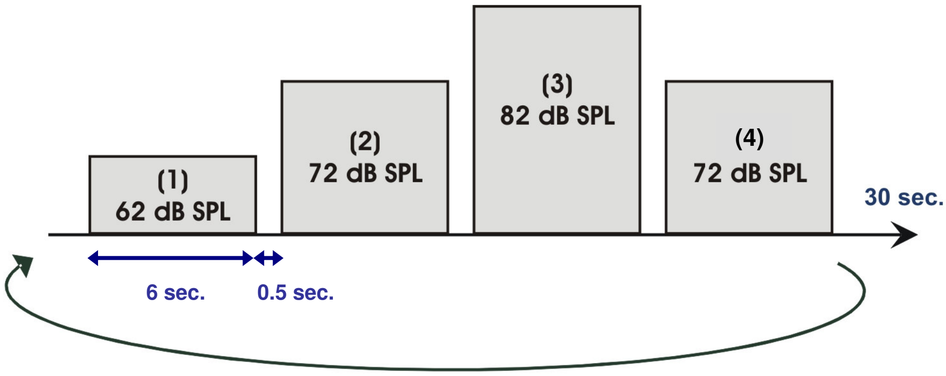

Figure 4.1. Schematic illustration of the format for input signals used in this study. All signals had a total duration of approximately 30 seconds and contained four segments of about 6 seconds each. In between each segments where was a pause of approx. 0.5 seconds (except in signal 4, the football match, which was a continuous signal).

4.2.1.1 Dantale speech & party versus car noise.

Signal (1) and (2) were made to exemplify electrically-reproduced speech, being presented at three levels in a noisy background. The two signals were identical in construction, with four speech and noise segments at a +5 dB signal-to-noise ratio. Only the noise type was different, being modulated party noise in signal (1) and a more static car cabin noise in signal (2).

Signal (1) represents a situation where speech, spoken at a normal vocal effort, is reproduced at higher levels. This would occur for instance in a movie theatre or when listening to a radio at a high volume setting. Signal (2) should represent a situation where the driver, listening to the car radio, needs to turn up the volume in order to compensate for the increased cabin noise when driving at higher speed, e.g., on the highway.

The speech parts used in both signals were taken from the Dantale-speech material (Elberling et al, 1989) - specifically the recording of a female speaker, reading from a text about the Danish island "Samsø" (Andersen, 1983). For the purpose of this study, four different segments of approximately 6 seconds were cut from the original recording. The sentences spoken in the four segments did not belong to the same context, but this was considered to be less important because speech intelligibility was not in focus for this study. The background noises used in signal (1) and (2) were taken from a compact disc containing environmental sound examples (Widex, 1999). A recording of party noise was used for signal (1), and a recording of car cabin noise was used for signal (2).

Long-term spectra for the Dantale speech and for the two noise signals are shown in figure 4.2. In this and in the following figures, spectra are shown as 1/3-octave band levels in dB, relative to full scale, which corresponds to a sample value of 34,000 in the sound editing program. The relative RMS level difference of 5 dB between the speech and noise are kept in the plot. It can be seen how the two noises differ in their spectral shape.

The party-noise has almost constant energy per band in the range from 100 Hz to 10 kHz, whereas the car-noise has most energy in the 100 Hz region and less energy at higher frequencies. The slope of the car-noise spectrum is approx. -13 dB per octave, above 300 Hz. The spectrum for the Dantale speech shows the two prominent peaks of the fundamental frequency (at 200 Hz) and the first formant (at 500 Hz).

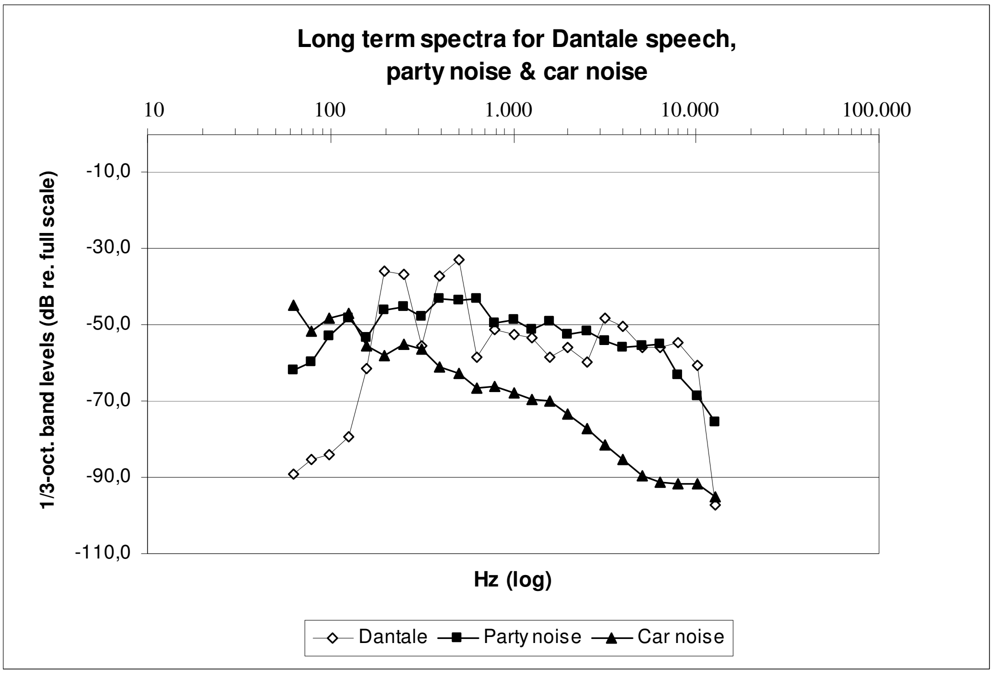

Figure 4.2. Long-term spectra (1/3-octave band levels) for the Dantale speech and for the two noise-signals used in signal (1) and (2). The SNR between speech and noise signals are +5 dB.

4.2.1.2 Speakers & party noise.

Signal (3) was made to resemble a real conversation between two people at a noisy party. In contrast to signal (1) and (2), the sentences in the four segments of this signal were spoken with different voice levels – that is, normal, raised and loud vocal efforts.

The speech parts for signal (3) were taken from a recording of a female and male speaker, reading from a text about a whale-expedition off the coast of Greenland (Widex, personal communication). In that recording, the voice levels of speakers were controlled to match the RMS-levels for normal, raised and loud vocal effort, as specified in the ANSI S3.75 standard (ANSI, 1997).

For the purpose of this study, the RMS-levels of the three vocal efforts were manipulated in the sound editing program to be 10 decibels apart. Thus, segment one contained speech at "normal vocal effort" at 62 dB SPL, segment two "raised vocal effort" at 72 dB SPL, segment three "loud vocal effort" at 82 dB SPL, and segment four "raised vocal effort" at 72 dB SPL. The party noise used in signal (1) was also used for this signal. Like in signal (1) and (2), the signal-to-noise ratio was +5 dB in each segment.

Long-term spectra for speech and party-noise in the 1st, 2nd and 3rd segments are shown in figures 4.3-4.5. In figures 3 and 4, a general increase in the speech energy can be seen in the 1-5 kHz region. This is the "Lombard effect" (see chapter 1), i.e., when increasing vocal effort in order to be heard over the noise, speech energy in the mid- to high-frequency region is increased more than at lower frequencies.

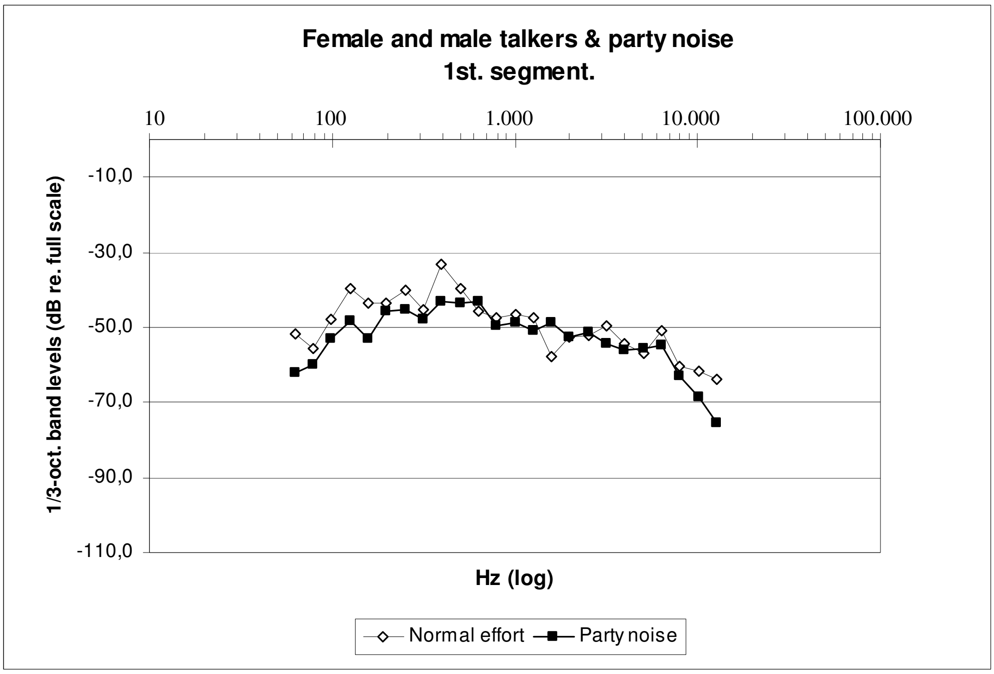

Figure 4.3. Long-term spectra (1/3-octave levels) for the 1. segment of signal (3), containing female and male talking & party noise. The SNR between speech and noise signals is +5 dB.

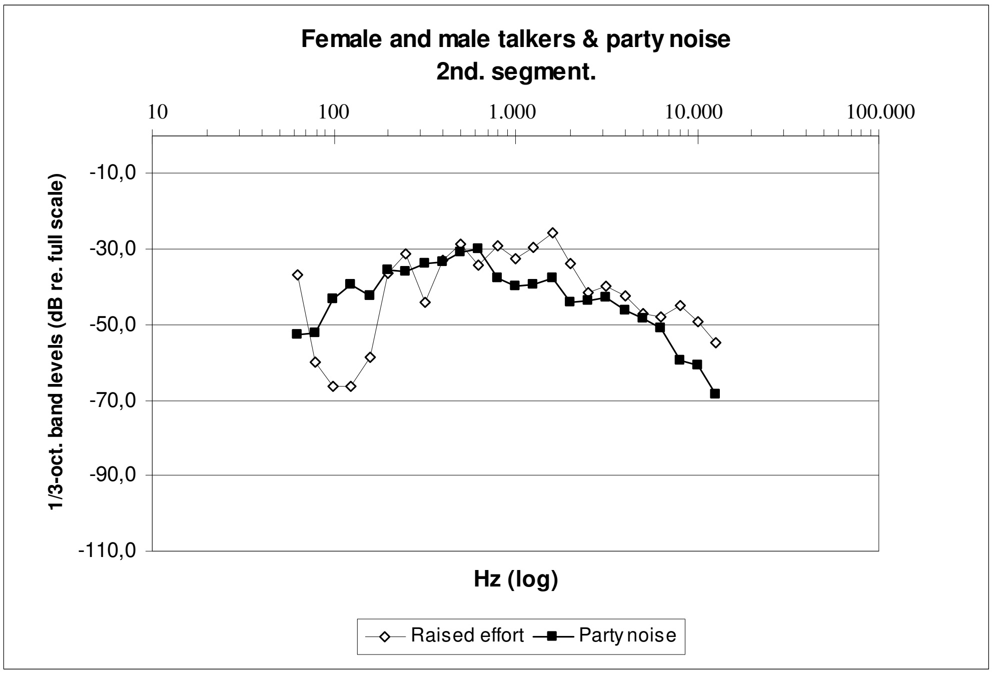

Figure 4.4. Long-term spectra (1/3-octave levels) for the 2. segment of signal (3), containing female and male talking & party noise. The SNR between speech and noise signals is +5 dB.

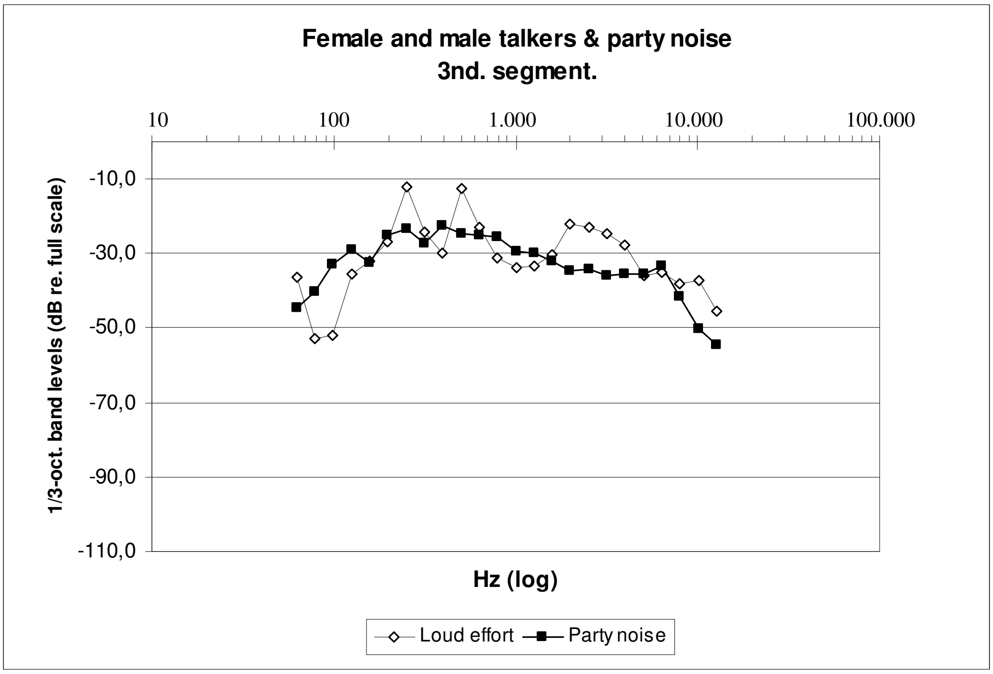

Figure 4.5. Long-term spectra (1/3-octave levels) for the 3. segment of signal (3), containing female and male talking & party noise. The SNR between speech and noise signals is +5 dB.

4.2.1.3. Audience at a football match.

Signal (4) was a purely environmental signal containing sounds and cheers from a football match. This signal was constructed from a 3-minute sound recording of a football match, with soft and loud passages (Oticon, 1993). Four different passages with overall RMS-levels of 62, 72, 82 and 72 dB SPL were mixed together, to form a natural continuous passage with four level segments. Long-term spectra of the Football match obtained from the 1^{st}, 2^{nd} and 3^{rd} segments are shown in figure 4.6. It can be seen that the spectra in the 2^{nd} and 3^{rd} segments have energy over a broader frequency range, compared to the spectrum in the 1^{st} segment.

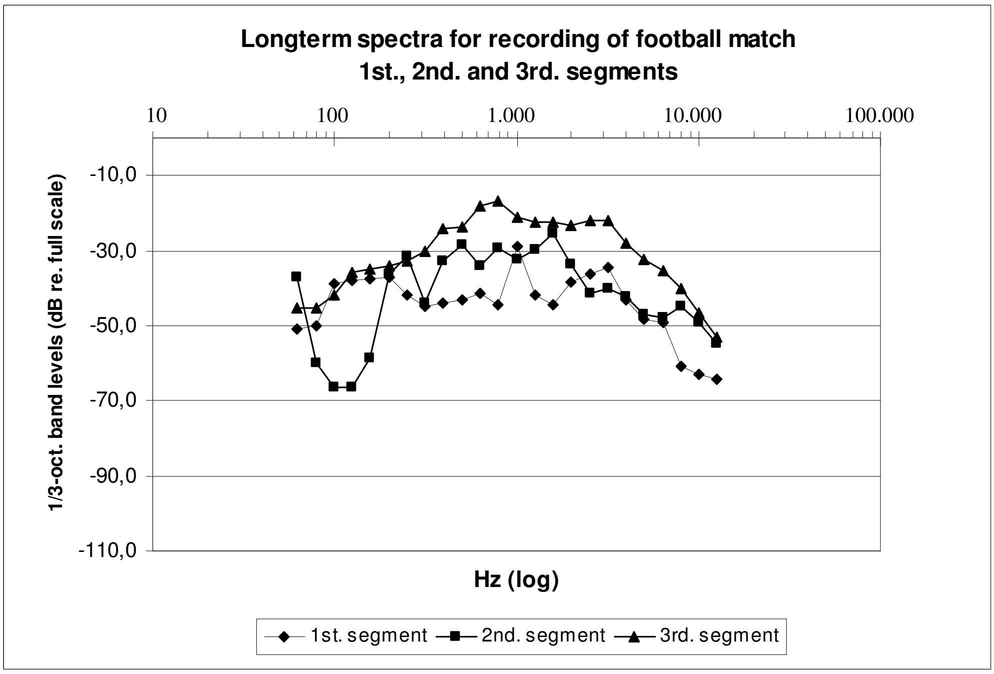

Figure 4.6. Long-term spectra (1/3-octave levels) of the 1^{st}, 2^{nd} and 3^{rd} segments of signal (4), containing sounds and cheers from a football match. The SNR between speech and noise signals is +5 dB.

In each of the four signals, the 3rd segment contained the highest RMS-level of the signal. This segment was therefore assumed to be the most challenging one for the test subjects, in regard to their tolerance for loud sounds. The long term spectra of the 3rd segment in each of the four test signals are shown for comparison in figure 4.7. It should be noted that the spectra shown for signal (1), (2) and (3) are of the combined speech and noise signals, mixed together at a +5 dB signal-noise-ratio.

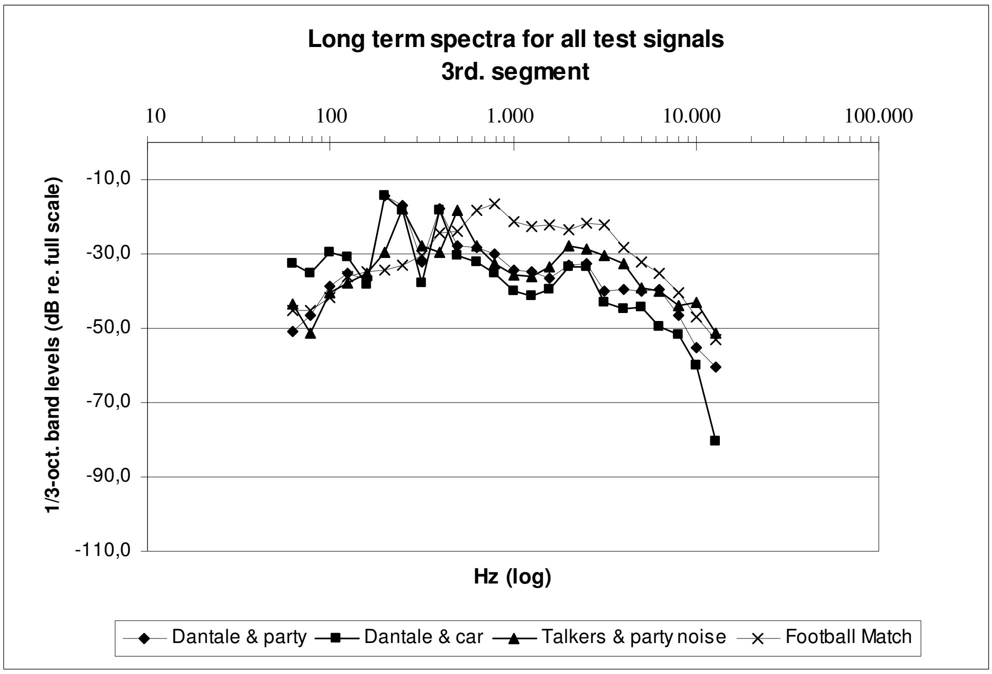

Figure 4.7. Long-term spectra (1/3-octave levels) of the 3rd segments in all four signals used in this experiment. The SNR between speech and noise signals are +5 dB.

From figure 4.7 it can be seen that the three speech and noise signals have fairly similar spectra. When comparing the two signals with Dantale speech, signal (1) containing party-noise has more energy above 12 kHz. On the contrary, signal (2) containing car-noise, has more energy below 150 Hz than all other signals. Signal (3) containing speech at loud vocal effort & party-noise has more energy in the 1100-1400 Hz region than signals (1) and (2). Finally, signal (4) containing the recording of audience at a football match, has more energy than all other signals above 5 kHz, but less energy than other signals in the 1500 – 3000 Hz region.

4.2.2 Compression of signals

The four input signals were compressed off-line in an experimental compressor with three independent compression-channels. The compressor was implemented as a Simulink-model in MATLAB by Carsten Paludan-Müller (personal communication, Nov. 1993). A schematic illustration of the model is shown in figure 4.8.

4.2.2.1 Description of compressor model

The compressor model can be described as having five stages: (1) the **input stage** where two mono-signals (i.e., speech and noise) are fed to the model at a given signal-to-noise ratio and summed together, (2) the **filterbank** where the signal is split into the three channels, (3) the **analytical stage** where the levels in each channel are detected and the compression parameters are specified, (4) the **summation stage** where the signals in each channel are multiplied by the factors specified in the analytical stage and the three channels are summed together, and finally (5) the **output stage**, where the path for the output file is specified.

The cross-over frequency between the 1st and 2nd channels was set at 833 Hz, and between the 2nd and 3rd channels at 2500 Hz. The dynamic properties of the compressor were tested and found in agreement with the IEC 118-0 standard (IEC, 1983).

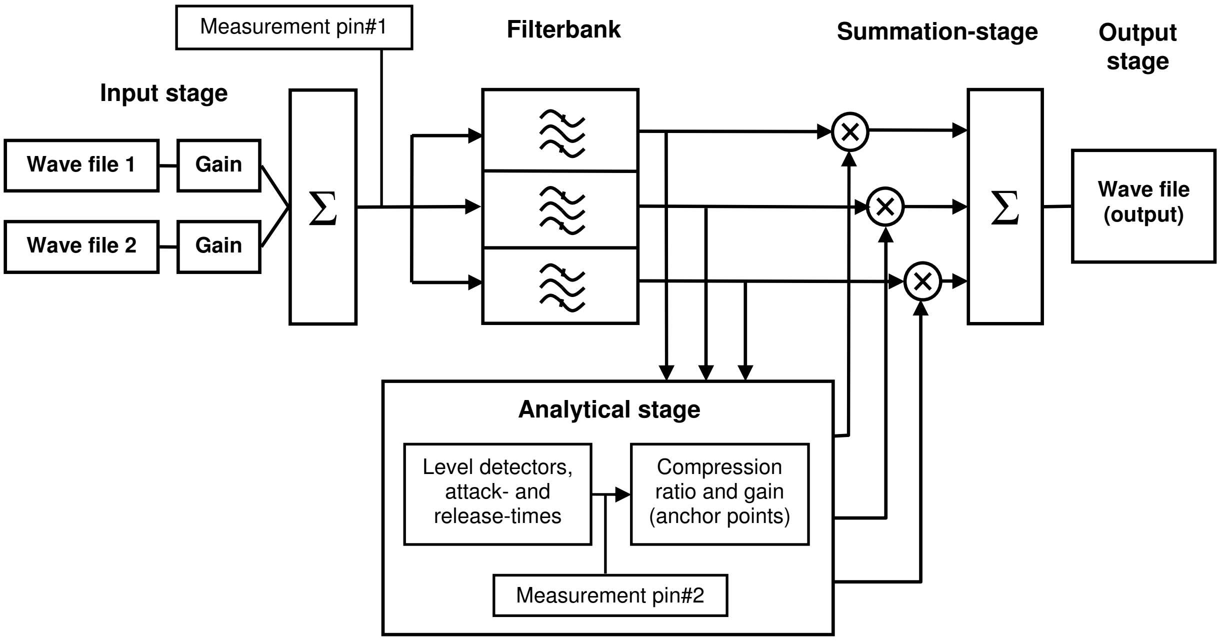

Figure 4.8. Illustration of the three-channel compressor, implemented in MATLAB by C. Palludan-Müller (. See text for details.

The analytical stage in the model is divided into two blocks; the **timing block** and the **gain block**. In the timing block, the attack- and release-times in each channel are specified. In the gain block, the compression ratio and the anchor-points are specified for each channel. The anchor-point in a given channel is the input level that receives the same gain in dB, regardless of the compression ratio.

Two measurement-pins were implemented in the model. The first pin was placed in-between the input-stage and the filterbank. This pin measures the overall RMS-level of the summed input signal before filtering takes place. The second pin was placed in-between the timing block and the gain block at the analytical stage. This pin measures the RMS-level in each channel, after the specified time constants had been applied. That is, the levels measured at the second pin reflect the influence of the attack and release-times on the level detectors in the timing block. These levels are the input to the gain block, in which the compression ratio and anchor-points are specified.

4.2.2.2 Calibration of the compressor and setting of compression parameters

Reference signals for the speech and noise signals (signal 1, 2 and 3), containing only speech (with pauses removed) from the 1st segment, were sent through the compressor model in the linear-setting (CR = 1:1). The input-gains were adjusted such that the RMS input level, measured at the first measurement pin, was 62 dB SPL (or -34 dB re. full scale) for each reference signal. In the timing block, the attack-time was set to 100 ms and the release-time to 5000 ms in each channel. Then, the anchor-points in the gain block were adjusted in each channel to the RMS-levels measured at the second measurement-pin (see appendix 9.2). This was done to maintain the spectral shape of the signal, according to the levels detected in the timing-block.

With this setup of the compressor, speech in the first segment would always receive the same overall gain regardless of the compression ratio. The second and fourth segments (72 dB SPL), and the third segment (82 dB SPL) would receive less and less gain with increasing compression ratio – diminishing the degree of level variation between the four segments in the signal.

Similarly, a reference signal was generated for signal (4), containing the sound of the football audience from the 1st. segment in that signal. Also for this signal, the gain was adjusted such that the first segment had an RMS input level of 62 dB SPL, and the anchor-points in the compressor block were adjusted to the levels measured after the timing block.

Each of the four signals were compressed with seven different compression ratios: 1:1 (linear condition), 1.25:1, 1.5:1, 2:1, 3:1, 5:1 and 10:1, giving a total of 28 test signals. The chosen compression ratios (except 1.25:1) resembled those used by Neuman et al (1995b), and gave audible differences among output signals, regarding the perceived degree of level variation. The choice of the attack time of 100 ms and release time of 5000 ms was made to simulate a slow acting compression scheme, which is comparable to the one used by Smeds (2004a). She used attack- and release-times of 2000 ms.

The broadband static input-output characteristics of the experimental compressor are shown in figure 4.9. The overall RMS input levels of the segments in each signal (the speech levels in signals (1)-(3) and the overall levels in signal (4)) are encircled on the abscissa.

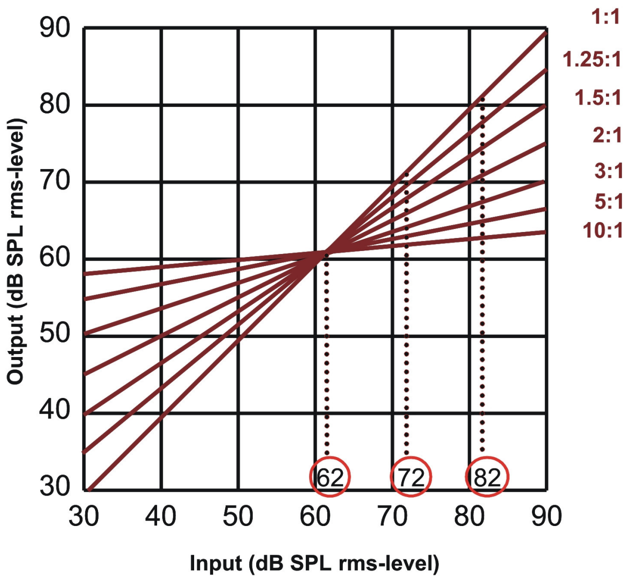

Figure 4.9. Illustration showing the broadband input-output characteristics of the experimental compressor. The overall RMS input levels of segment 1. (62 dB SPL), segment 2. and 4. (72 dB SPL) and segment 3. (82 dB SPL) in each signal are encircled on the abscissa.

Sound examples of all compressed signals can be found on the audio-CD in appendix 9.1. In figure 4.10, waveforms of the seven compressed versions of the *Dantale & party noise* signal are shown. Note that in the version of this signal that was compressed with a ratio of 10:1, very little level difference exist between the four segments. The measured difference in RMS-level was on the order of 1-2 dB.

Figure 4.10. Waveforms of the processed versions of the *Dantale & party noise* signal, compressed with ratios from 1:1 (top panel) to 10:1 (bottom-panel) (Screen capture from sound editing software).

4.2.3 Test setup for listening experiment #1

4.2.3.1 Presentation of test signals

The compressed signals were presented to listeners in the anechoic chamber at Ørsted●DTU. Signals were played back using the sound-editing program (Adobe Audition, version 1.0), running on a laptop computer. The signal was routed through a 24 bit soundcard (Creative Estigy) to the loudspeaker amplifier (QUAD 606). A Rogers LS3/5A loudspeaker was used for presenting the test signals. This loudspeaker has a power handling of 25 Watt and a frequency response of +/- 3 dB in the range from 70 Hz – 20 kHz (see appendix 9.3).

4.2.3.2 Calibration of presentation levels

Test subjects were seated at three meters distance from the loudspeaker. White noise with a bandwidth of 750 Hz, centred at 1 kHz, was used to calibrate the presentation level at the position of the listener. The noise was given a sound level equal to the RMS-level of speech in the 1st segment of the compressed signals. In signal (4), the overall level of the 1. segment was used as reference.

A Brüel & Kjær sound level meter, type 2240 was used to calibrate the test setup. Before each listening test, the calibration signal referring to the signal being tested was adjusted to 62 dB SPL (+/- 1 dB) at the position of the listener. In this way, speech in the first segment of signals (1), (2) and (3) and the sounds in the 1st segment of signal (4) would have an RMS-level of 62 dB SPL at the position of the listener.

4.2.3.3 Hearing aids worn by test-subjects and the NAL-R fitting-rationale

All subjects wore binaural BTE hearing aids (Widex Senso Diva, see appendix 9.4), fitted linearly according to the National Acoustic Laboratories Revised (NAL-R) fitting procedure (Byrne & Dillon, 1986). Hearing aids were fitted with custom-made earmoulds, each with a 1.2 mm ventilation channel. The purpose of the hearing aids was only to amplify the compressed signals, placing speech of 62 dB SPL at the most comfortable levels of the individual listeners. In this way, the combination of the compressed signals presented from the loudspeaker and the linear gain in the hearing aids, would simulate a non-linear hearing aid.

The objective of the NAL-procedure is to present average speech at the most comfortable level, such that all frequency areas contribute equally to the overall loudness of the signal (denoted *loudness equalisation*). The first version of this procedure prescribed gain to be 0.46 times the loss at 1 kHz, but with corrections at all other frequencies according to the *60 phon equal loudness contour* in normal listeners and the *long term average spectrum of speech* (Byrne & Tonisson, 1976).

The revised NAL-procedure from 1986 was developed because the original NAL-formula did not meet its aim of providing equal loudness, especially at low frequencies. Experimentally determined responses were related to the subjects' audiometric configurations. A new formula was derived (figure 4.11), that preserves the half-gain rule (0.46) for the average of thresholds at 500, 1000 and 2000 Hz (3FA), but in addition contained a slope rule that varies the slope of the response as a function of the variation in the audiogram slope.

1. Calculate $X = 0.05\,(H_{500} + H_{1k} + H_{2k})^{a}$

2.
$$
\begin{aligned}
G_{250} &= X + 0.31\,H_{250} - 17^{b}\\
G_{500} &= X + 0.31\,H_{500} - 8\\
G_{750} &= X + 0.31\,H_{750} - 3\\
G_{1k} &= X + 0.31\,H_{1k} + 1\\
G_{1.5k} &= X + 0.31\,H_{1.5k} + 1\\
G_{2k} &= X + 0.31\,H_{2k} - 1\\
G_{3k} &= X + 0.31\,H_{3k} - 2\\
G_{4k} &= X + 0.31\,H_{4k} - 2\\
G_{6k} &= X + 0.31\,H_{6k} - 2
\end{aligned}
$$

[a] *H, HTL (ISO standard).*
[b] *G, insertion gain.*

Figure 4.11. The revised NAL-formula, prescribing targets for real-ear gain. X equals the ratio of the new and old gain requirements (0.46/0.31=0.05, times the 3FA). 0.31xH is the variation in the response slope as a function of audiogram slope. The final constant determines the variation in the response as a function of frequency (Byrne & Dillon, 1986).

The NAL-R procedure was chosen for this study, because it has been thoroughly validated and found to achieve its aim of providing good listening comfort and speech intelligibility in the majority of users (Byrne & Cotton, 1988). This procedure has also gained wide clinical acceptance, and has been used as a reference in several studies investigating compression in hearing aids - including the studies by Neuman (1995b) and Hansen (2002).

NAL-R targets for insertion gain were calculated from individual hearing thresholds at 500, 1000, 2000 and 4000 Hz. Targets were then inserted in the fine tuning screen of the software used for fitting the Senso Diva Hearing aid (Widex Compass, version 3.4.1). Same values were used for IGnormal, IGloud and IGsoft, such that the hearing aid provided the same amount of gain at all input-levels (i.e., linear gain).

The fitting screen, with insertion-gain targets for one of the participants in the study, is shown in figure 4.12.

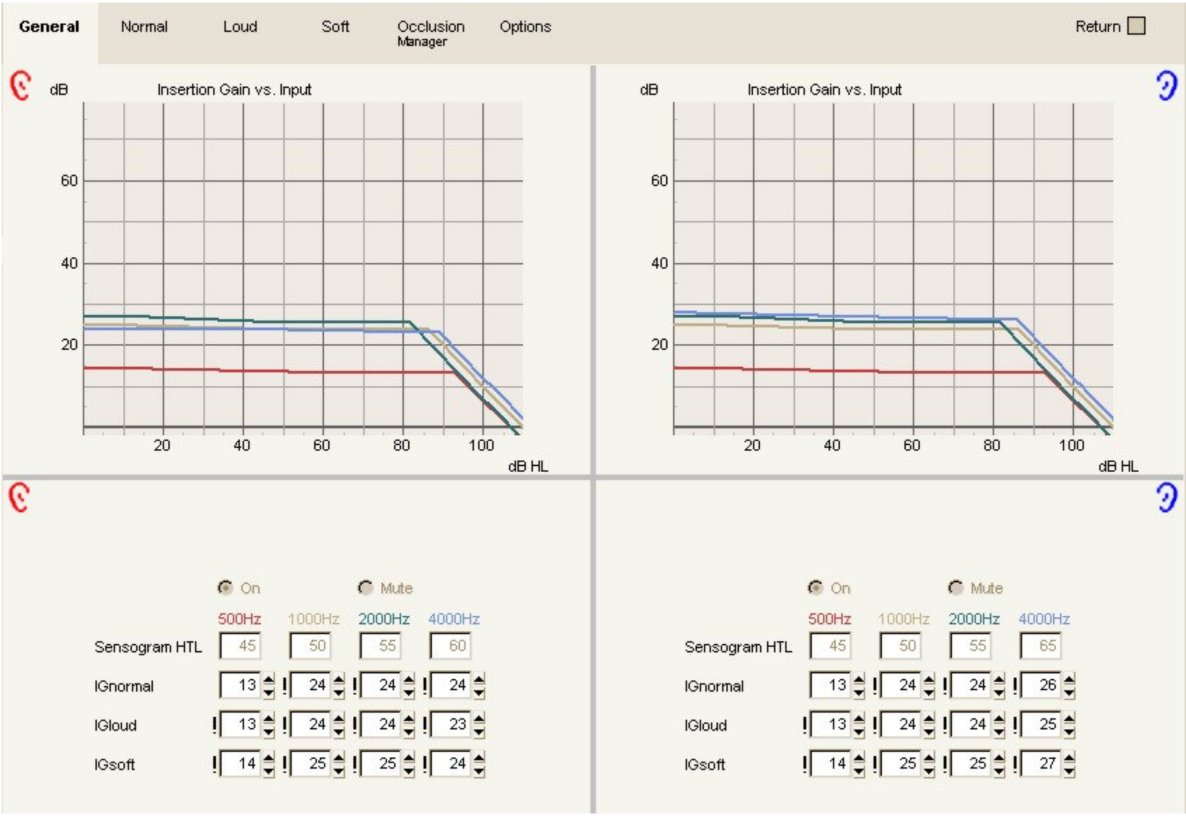

Figure 4.12. Fine tuning screen from the fitting software.
Insertion gain values are shown for one of the test subjects in the experiment.

In the options-setting, the microphone was set to fixed omni-directional mode. The automatic output control (AOC) was turned on. This was done to avoid distortion in the hearing aid output at very high input levels that might negatively affect a listener's sensation of the test signals. The fitting data for all subjects, together with test box-measurements of gain- and output, can be found in appendix 9.5. A picture of the test setup in the anechoic chamber is shown in figure 4.13.

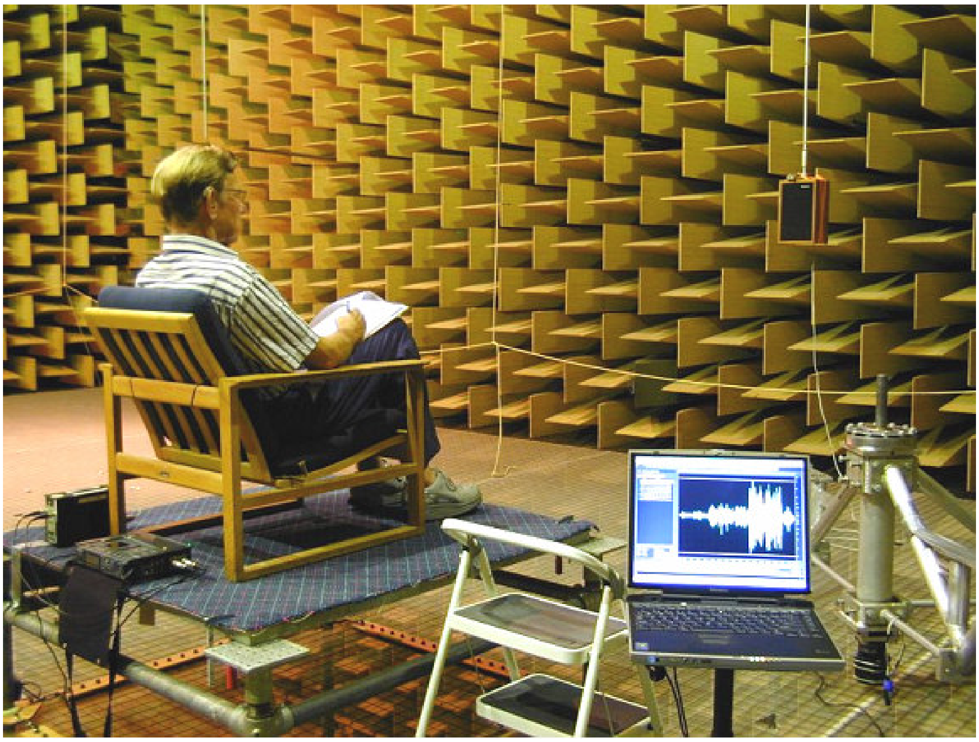

Figure 4.13. Test subject listening to compressed signals in the anechoic chamber. All subjects wore binaural BTE hearing aids, fitted linearly to NAL-R for the individual hearing loss – amplifying the compressed signals presented from the loudspeaker.

4.2.4 Description of test-subjects

4.2.4.1 Audiometric configurations

Eight hearing-impaired listeners with moderately sloping losses participated in the study. They were four males and four females ranging from 60 – 85 years of age, with a mean age of 73.5 years. Subjects were tested to have normal middle ear function. Thresholds configurations for the right and left ears of all subjects are shown in figure 4.14.

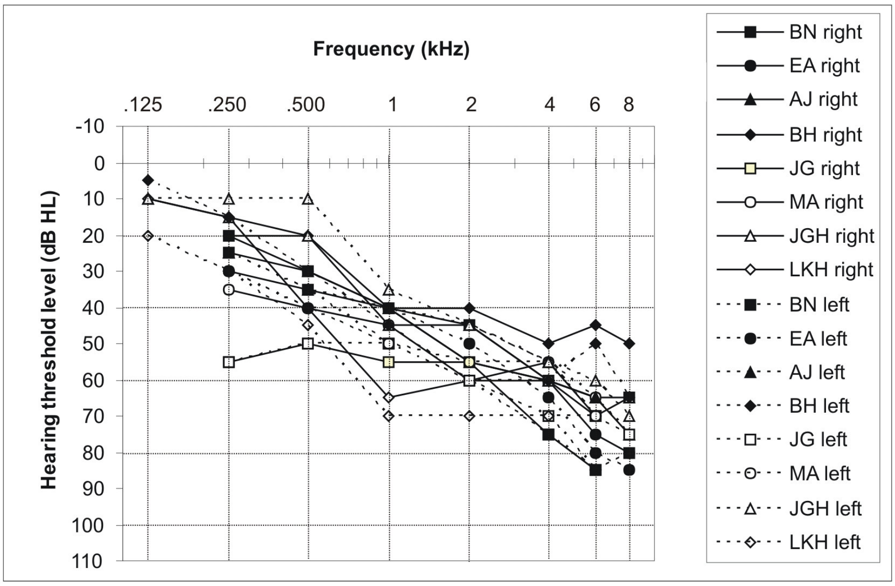

Figure 4.14. Thresholds configurations for right and left ears of the eight subjects in the study.

4.2.4.2 Previous hearing aid experience

All subjects had been using hearing aids for between seven and twenty-four years and could be considered as experienced users. They were all currently using digital hearing aids with automatic gain regulation from different manufacturers. All subjects indicated that they wore hearing aids on both ears for most of the day.

4.2.4.3 Preferences regarding loud sounds

Six of the subjects had a volume control on their personal device, allowing them to override the automatic gain adjustments. The last two subjects had no volume control on their device. All subjects were asked to fill out a short questionnaire concerning their use of the volume control in nine different listening situations (see appendix 9.6). The chosen situations were all likely to contain mid to high sound pressure levels. The two subjects without volume controls on their device were asked to imagine, what they would do in the various situations. Seven subjects returned the questionnaire. The number of subjects turning the volume control up or down in each situation, are shown in figure 4.15.

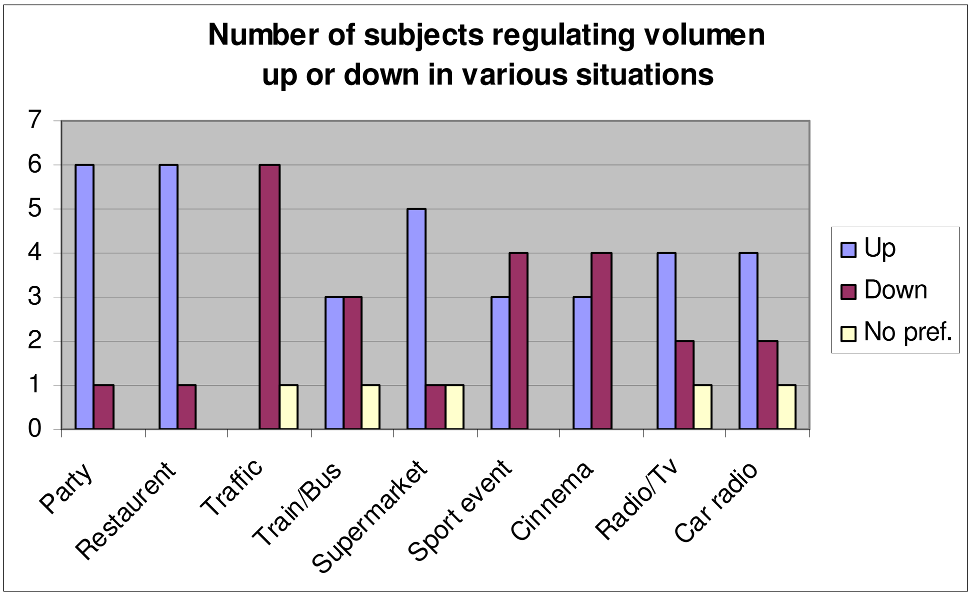

Figure 4.15. **Number of subjects turning the volume control up or down, in nine different listening situations.**

In three of the situations, *being at a party*, *in a restaurant* and *at the supermarket*, there was a clear preference for turning up the volume. Subjects indicated that they needed to turn up the volume, in order to understand speech better in these noisy conditions. Similar trend was seen in the two situations, *listening to the radio/TV* and *car radio*.

On the contrary, in traffic situations, most subjects preferred to turn down the volume, as this situation contained only annoying sound. In the remaining three situations, there was an equal preference for turning the volume either up or down. This small survey shows that listeners in this study are aware of the loudness and speech intelligibility in different situations, and that they try to adjust their hearing aids to some preferred listening level.

4.2.5 Procedure for listening experiment #1

The seven compressed versions of each input signal were presented four times to each subject, giving a total of 28 presentations. The 28 test signals were randomized for each subject.

The first 7 presentations acted as training. Subjects rated each presentation on three categorical-scales (discussed below). Test signals were presented during four sessions (on different days) of approximately one hour each, with signals belonging to a specific input signal in each session. Table 4.1 provides an overview of the four test sessions.

Table 4.1. Overview of the four test sessions in experiment #1.

Session 1	Session 2	Session 3	Session 4
Dantale & party noise	*Dantale & car noise*	*Speakers & party noise*	*Football Match*
7 compressed versions of signal (1), presented 4 times = 28 presentations (randomized for each subject). The first 7 presentations acted as training.	7 compressed versions of signal (2), presented 4 times = 28 presentations (randomized for each subject). The first 7 presentations acted as training.	7 compressed versions of signal (3), presented 4 times = 28 presentations (randomized for each subject). The first 7 presentations acted as training.	7 compressed versions of signal (4), presented 4 times = 28 presentations (randomized for each subject). The first 7 presentations acted as training.

4.2.5.1 Subject instruction and rating on categorical scales

Subjects were given both written and oral instructions about the purpose of the experiment (see appendix 9.7). It was explained that the test investigated how hearing aid users perceive level variations processed through a hearing aid, and what degree of level variation they prefer. Before the test started, the original uncompressed signal with the full degree of level variation and the processed version of the same signal, compressed with a 10:1 ratio, was presented to them. They were told that the degree of level variation in the following signals would be within this range.

During the actual test, each compressed signal was presented in loops while the listener made their ratings on three psychometric scales with a pencil (see fig. 4.16 and appendix 9.8). The construction of these scales was similar to the ones used by Gabrielsson et al. (1979, 1985, 1990) for subjective sound quality measures (see fig. 2.9). Each scale had 10 major marks, divided into 100 minor marks, with five verbal categories positioned at the 1st, 3rd, 5th, 7th and 9th major marks. Subjects were told to use the whole scale, also the intervals between the main categories if they found this necessary.

On the first scale (**variation**), listeners were asked to mark the *degree of level variation in the four segments* they perceived in the given recording. The scale contained the categories; "No variation", "Small variation", "Midway", "Large variation" and "Very large variation". The purpose of this scale was to measure the variation perceived by listeners in relation to the given signal and the applied compression ratio.

On the second scale (**loudness**), listeners were asked to mark the loudness of the three first segments of each signal, with the numbers 1, 2 and 3. This scale contained the main categories; "Not audible", "Very soft", "Comfortable", "Loud", "Very loud" and "Uncomfortably loud". The purpose of this scale was to obtain information about the upper part of the listener's dynamic range and the difference in loudness between segments, in relation to the compression ratio used.

Finally, on the third scale (**acceptance**), listeners were asked to mark how acceptable the reproduction of the three levels would be, if they encountered this in a real listening situation. The main categories of this scale were "Highly unacceptable", "Unacceptable", "Tolerable", "Acceptable" and "Highly acceptable".

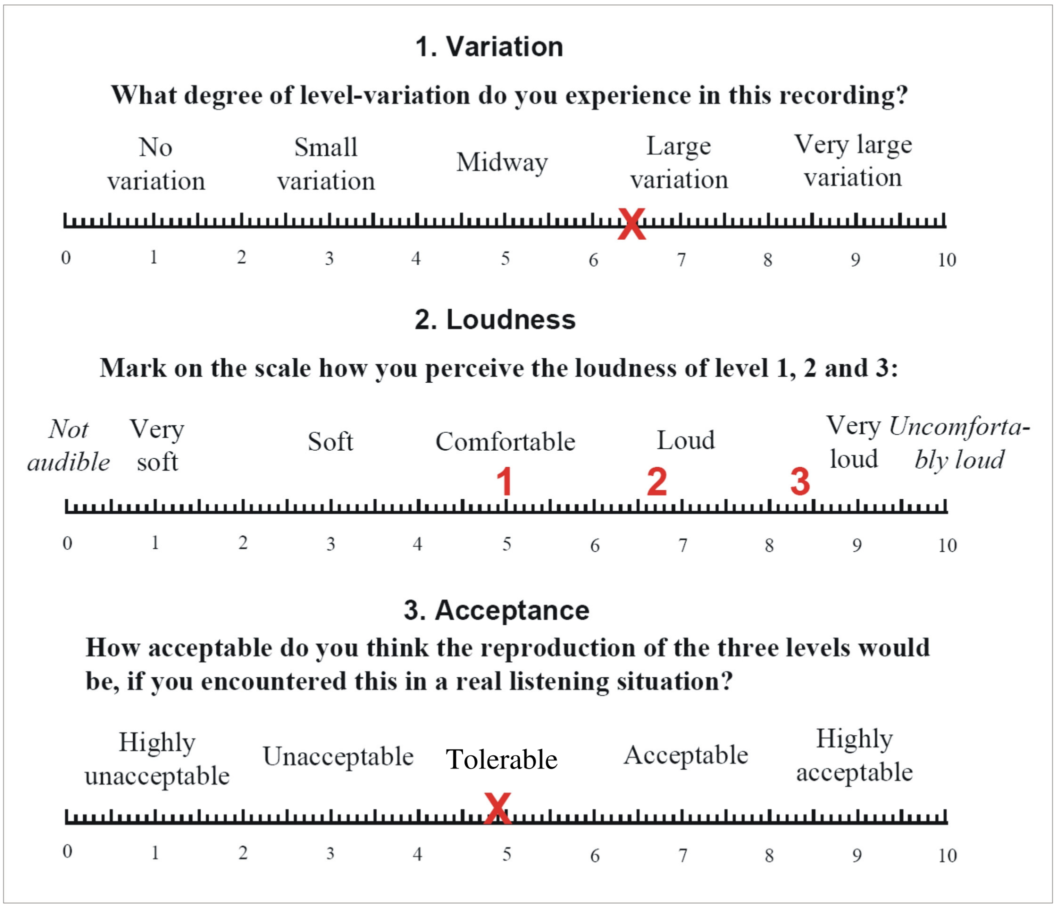

Figure 4.16. **The three categorical scales used for the subjective rating of variation, loudness and acceptance (English translation).**

The term *acceptance* may be regarded as related to the concepts of *listening comfort* and *the preferred listening level*. Listeners were informed that the goal was to provide them with some degree of the variation in the input signal, although the sound should not become unpleasantly loud for them. They were instructed to rate the overall acceptance of the level variation in the signal. Thus, this scale would provide information about the acceptance of the perceived level variation, in the given input signal at a given compression ratio.

Subjects rated the signals one at the time, and they could not see their ratings made for the previous signals. All ratings made by each subject (that is, 7x3 in each session – excluding ratings made in the training presentations) were collected and entered into a data spreadsheet for further analysis.

4.3 Results

The means of all subjects' ratings on each scale, as a function of compression ratio, are shown in figures 4.17, 4.18(a-c) and 4.19. In each graph the error-bars indicate the 95 % confidence interval for the given mean. Mean ratings are shown for each of the four input signals, which in the following are termed: **(1)** *Dantale & party*-noise, **(2)** *Dantale & car*-noise, **(3)** *Speakers & party*-noise and **(4)** *Football Match*.

4.3.1 Mean ratings of variation

Figure 4.17 shows the mean ratings of the perceived level variation in each signal, as a function of compression ratio.

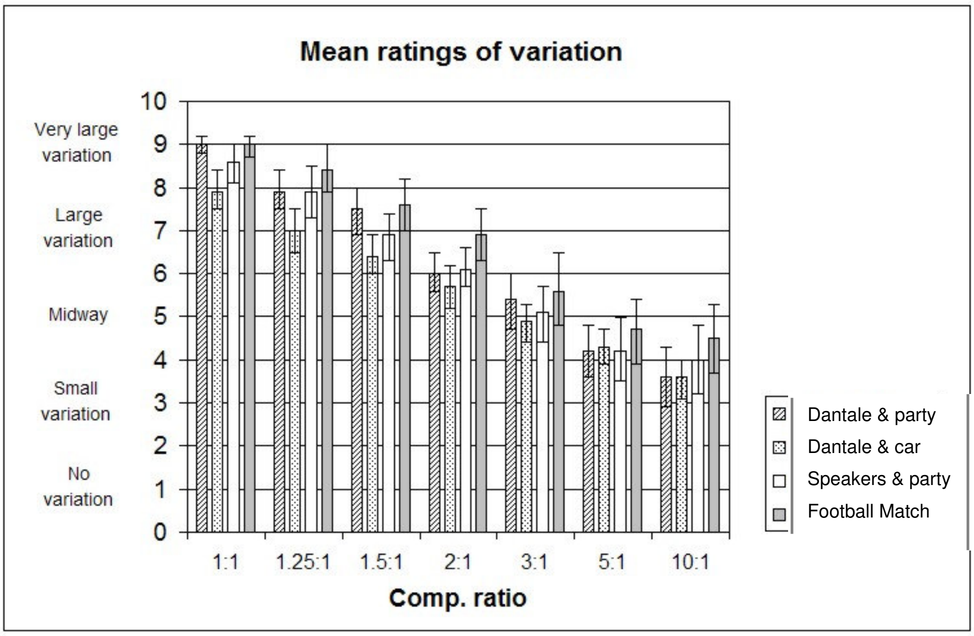

Figure 4.17. Mean ratings of variation in the four signals, as a function of compression ratio. Error-bars indicate the 95 % confidence interval for the given mean.

As expected, the degree of perceived variation diminishes for all signals, with increasing compression ratio. In the 1:1 condition, the ratings are in the region of 8-9 (approaching the "very large variation" category), whereas at 10:1 ratings are close to 4 (in-between the "small variation" and "midway"-categories). Overall, a signal effect is seen especially at the lower compression ratios; the *Football match* receives the highest ratings. Then follow the *Dantale & party*-noise and *Speakers & party*-noise signals, which receive almost equal ratings at all ratios. The *Dantale & car*-noise signal receives the lowest ratings. A statistical analysis of differences between ratings within same compression ratios is given later.

4.3.2 Mean ratings of loudness

Mean ratings of the perceived loudness of segments 1, 2 and 3 of each signal as a function of compression ratio, are shown in figure 4.18(a-c).

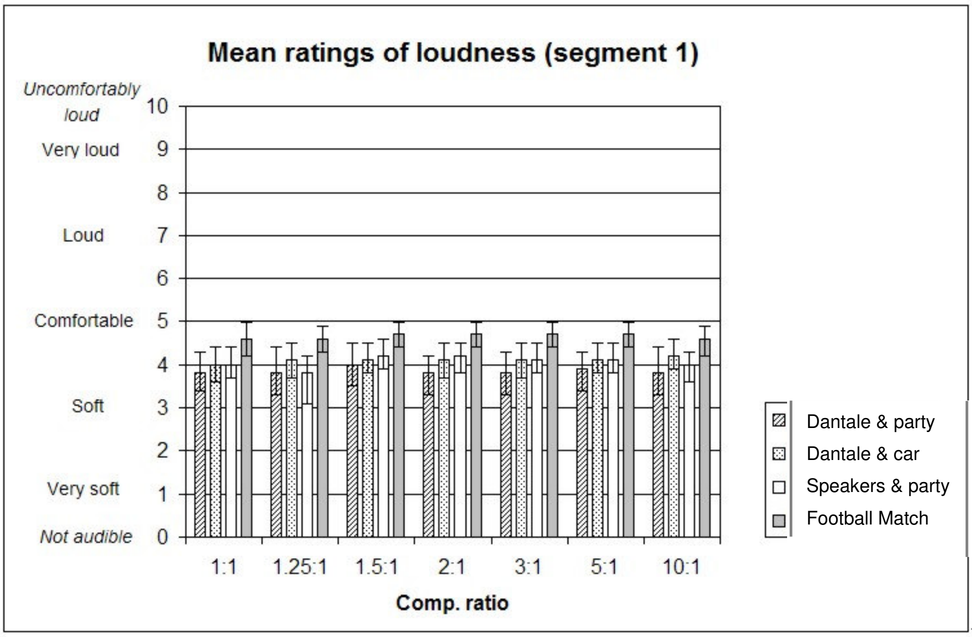

Figure 4.18a. Mean ratings of loudness (segment 1) in the four signals, as a function of compression ratio. Error-bars indicate the 95 % confidence interval for the given mean.

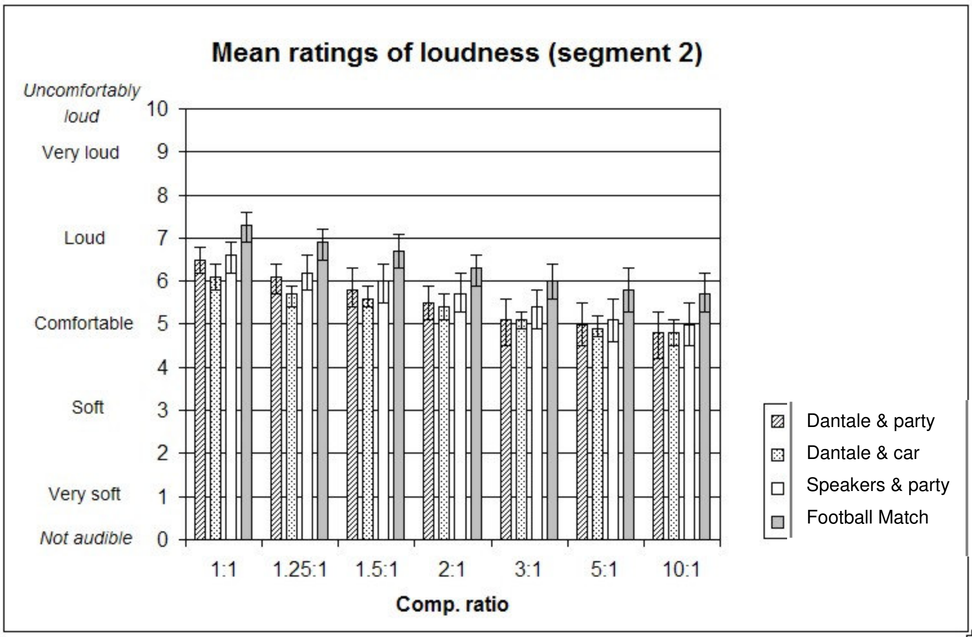

Figure 4.18b. Mean ratings of loudness (segment 2) in the four signals, as a function of compression ratio. Error-bars indicate the 95 % confidence interval for the given mean.

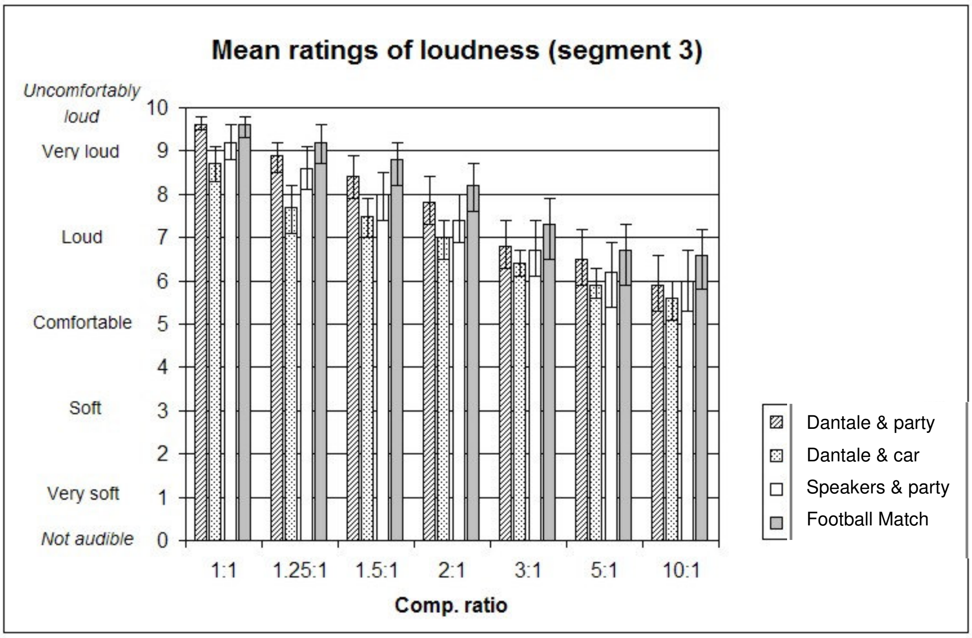

Figure 4.18c. Mean ratings of loudness (segment 3) in the four signals, as a function of compression ratio. Error-bars indicate the 95 % confidence interval for the given mean.

The differences in perceived loudness between the three segments diminish, as the compression ratio is increased – which is expected. The highest ratings are given to the 3rd segments (82 dB SPL) in the 1:1-condition. Here, the ratings are in the region from 8-10 (the categories “very loud” and “uncomfortably loud”). The lowest ratings are given to the 1st segment (62 dB SPL). Note that the loudness ratings of the first segment do not change within same signals, but stays around 4 (midway between the categories “soft” and “comfortable”).

Overall, the *football match* receives the highest ratings of loudness in all three segments, compared to other signals. In the 3rd segment there is a clear signal-effect, where ratings for the *football match* are followed by the *Dantale & party*-noise, then the *Speakers & party*-noise, and finally the *Dantale & car*-noise receiving the lowest ratings.

4.3.3 Mean ratings of acceptance

Mean ratings of acceptance in each signal as a function of compression ratio, are shown in figure 4.19. To recall, subjects were asked to indicate how acceptable the level variations in a given signal would be, if encountered in a real listening situation.

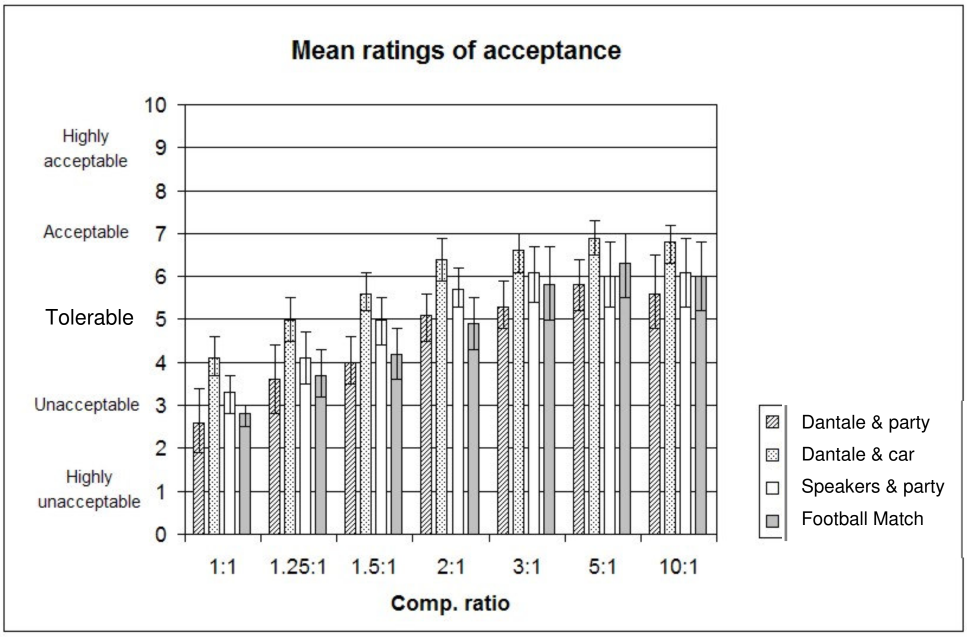

Figure 4.19. Mean ratings of acceptance across subjects for the four signals. Error-bars indicate the 95 % confidence interval for the given mean.

As seen in figure 4.19, the ratings increase with increasing compression ratio, to a point where they seem to stay within the same region. The lowest ratings are in the range from 2-4 ("unacceptable") at the 1:1-condition. At the highest compression ratios (3:1 – 10:1) ratings are in the range from 5-7 (in-between "tolerable" and "acceptable"). The *Dantale & car*-noise receives the highest degree of acceptance at all ratios, followed by the *Speakers & party*-noise. The *Football match* and the *Dantale & party*-noise receive the lowest ratings at almost every compression ratio.

4.4 Discussion of experiment #1

The gradually diminishing ratings of variation (figure 4.17), shows that listeners are able to perceive the difference in level variation between segments, when signals are processed with different compression ratios. In the 1:1-condition, the level interval of 20 dB between segment 1 and 3 was assessed by subjects to be a very large level variation. At a ratio of 10:1, subject still perceived a small level variation between the segments.

There may be two reasons for this: First, even though signals were compressed with a ratio of 10:1, where was still a difference in RMS-levels between segments of 1-2 dB that may have been audible to the listeners. Second, in the transition from the 1st to 2nd and from the 2nd to the 3rd segments in all signals, the compressor produced an audible *overshoot*. That is, too much gain was applied at the beginning of the segments, before it was turned down within the 100 ms attack-time. In addition, spectral differences between segments in the realistic signals

(3) *Speakers & party*-noise and (4) *Football match*, may also contribute to the sensation of level-differences at the highest ratios.

The ratings of loudness seem to be reflected in the ratings of level variation. That is, the loudness ratings of segments 1, 2, and 3 do not become equal at the higher compression ratios. In the 1:1-condition, the 3rd segment in all signals is perceived as being "very loud" or "uncomfortably loud". This is expected, as the linear gain in the hearing aids worn by subjects would provide a greater loudness in this condition, compared to the loudness of the same presentation level in normal-hearing listeners (see subsection 3.1.2).

In figure 4.18c, it can be seen that the ratings of the first segment are within the same range at all compression ratios. This confirms that the hearing aids (fitted to NAL-R) provided a comfortable loudness for speech at 62 dB SPL RMS-level – which was intended.

The pattern seen in fig. 4.19 of increasing acceptance at the higher compression ratios are in contradiction with findings in the literature, showing a preference for ratios no greater than 3:1 (Neuman et al, 1995a, 1998; Hansen 2002). This may partly be related to the slow time constants used in this experiment (AT = 100 ms and RT = 5000 ms). In the earlier studies, a decline in perceived sound quality, clarity of speech etc. was related to shortening of the release-time. This is discussed further below and in chapter 5.

4.4.1 Statistical comparison of signals within same compression ratios.

The statistical analysis was partly based on the description by Gabrielsson (1979b). On each of the three scales, a mixed model analysis of variance was carried out, using statistical software (SPSS, version 11.5). On the loudness scale, three independent ANOVA's were carried out on the ratings for the 1st, 2nd and 3rd segments. The fixed effects for the dependent variable RATING in each scale are shown in tables 4.2 to 4.6. The full data-output from the ANOVA for each scale together with model verification, can be found in appendix 9.10.

Table 4.2 Variation-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7,000	409,820	,000
TRIAL	2	653	,879	,416
CR	6	653	264,063	,000
SIGNAL	3	653	24,859	,000

Table 4.3 Loudness, 1 seg., tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7,000	221,836	,000
TRIAL	2	653	1,971	,140
CR	6	653	,598	,732
SIGNAL	3	653	66,112	,000

Table 4.4 Loudness, 2 seg., tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7	436,693	,000
TRIAL	2	653,000	1,861	,156
CR	6	653,000	92,570	,000
SIGNAL	3	653,000	100,534	,000

Table 4.5 Loudness, 3 seg., tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7	534,073	,000
TRIAL	2	653,000	,138	,871
CR	6	653,000	211,234	,000
SIGNAL	3	653,000	53,296	,000

Table 4.6 Acceptance-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7,000	411,874	,000
TRIAL	2	653	,060	,942
CR	6	653	61,353	,000
SIGNAL	3	653	27,120	,000

a = Dependent Variable: RATING.

In all scales, a significant effect of the factors SIGNAL and CR (compression ratio) was found – except for CR in the loudness-scale, 1st segment. No significant effect was found for the TRIAL-factor, which suggests that the procedure for randomisation of test signals eliminated any training- or order-effects.

A post hoc comparison with Bonferroni-correction was made to find significant differences in ratings ($p < 0.05$) of signals being processed with the same compression ratio.

Some comparisons are of special interest. The ratings of signal (1) *Dantale & party*-noise were compared to signal (2) *Dantale & car*-noise. These two signals had equal RMS-levels for speech, but differed in regard to the background noise (see fig. 4.2). A comparison of the mean ratings made for these two signals on the three scales, are shown in figure 4.20.

At the lower compression ratios, the *Dantale & party*-noise received significantly higher ratings on the variation scale and loudness scale (3rd segment), compared to the *Dantale & car*-noise. On the other hand the latter signal received significantly higher ratings of acceptance at all ratios.

The ratings of signal (1) *Dantale & party*-noise were also compared to ratings of signal (3) *Speakers & party*-noise. These two signals differed in the type of speech signal - being normal vocal effort reproduced at higher levels in signal (1), whereas in signal (3) speakers were talking with normal, raised and loud vocal efforts. A comparison was made to see if the difference in speech spectra would make a significant difference in ratings. In most cases no significant difference in ratings was found between the two signals, as seen in figure 4.21.

Finally, the two "realistic" signals (3) *Speakers & party*-noise and (4) the *Football match* were compared. In this comparison significant differences were found mainly in the loudness ratings, where the *Football match* received higher ratings at all compression ratios. In the variation and acceptance scales, no difference was found in most cases (fig. 4.22).

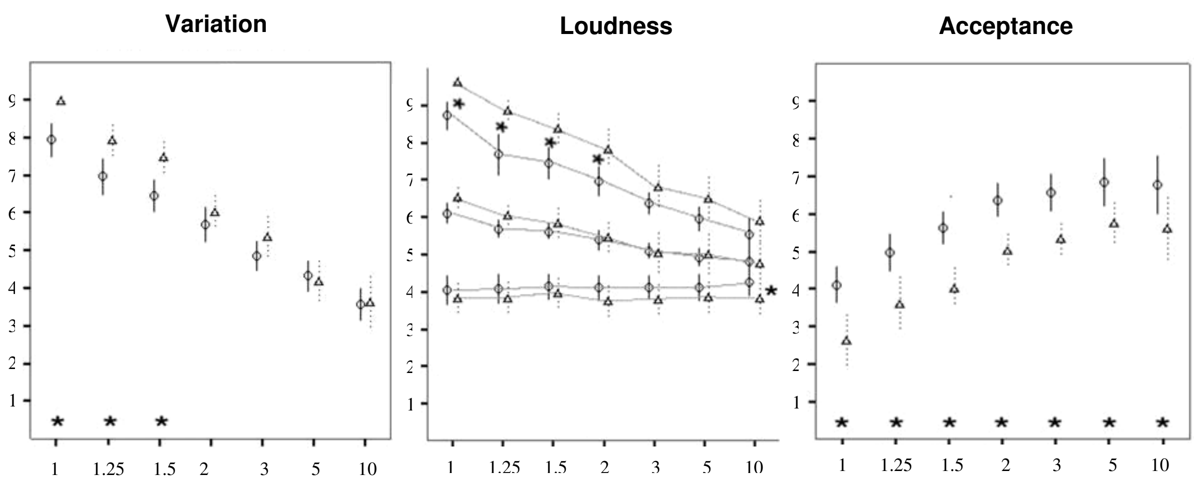

Figure 4.20. Comparison of ratings made for the *Dantale & party*-noise (Δ) and *Dantale & car*-noise (○) signals. (*) indicate a significant difference between ratings at the given compression ratio ($p < 0.05$). Vertical lines indicate the 95 %-confidence interval for the given mean.

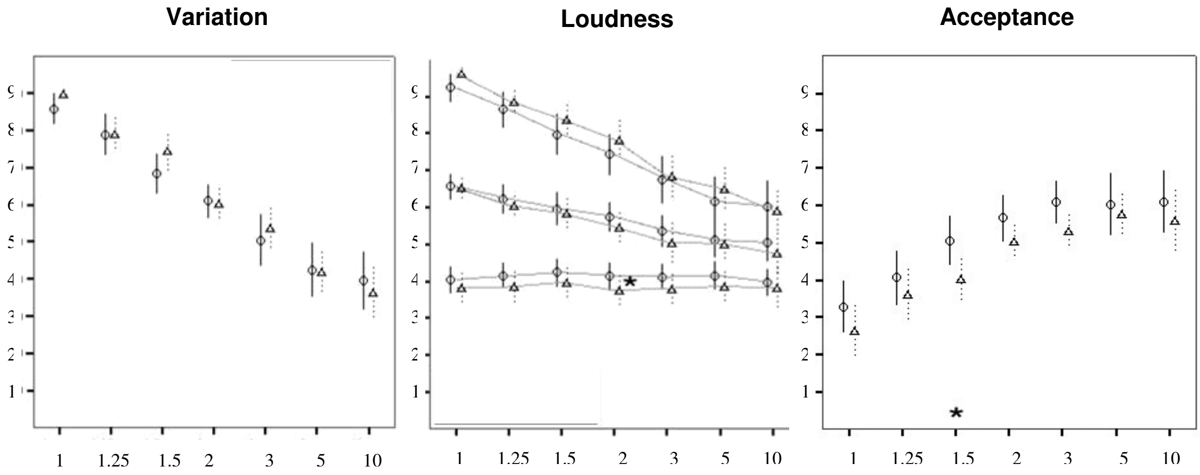

Figure 4.21. Comparison of ratings made for the *Dantale & party*-noise (Δ) and *Speakers & party*-noise (○) signals. (∗) indicate a significant difference between ratings at the given compression ratio ($p < 0.05$). Vertical lines indicate the 95 %-confidence interval for the given mean.

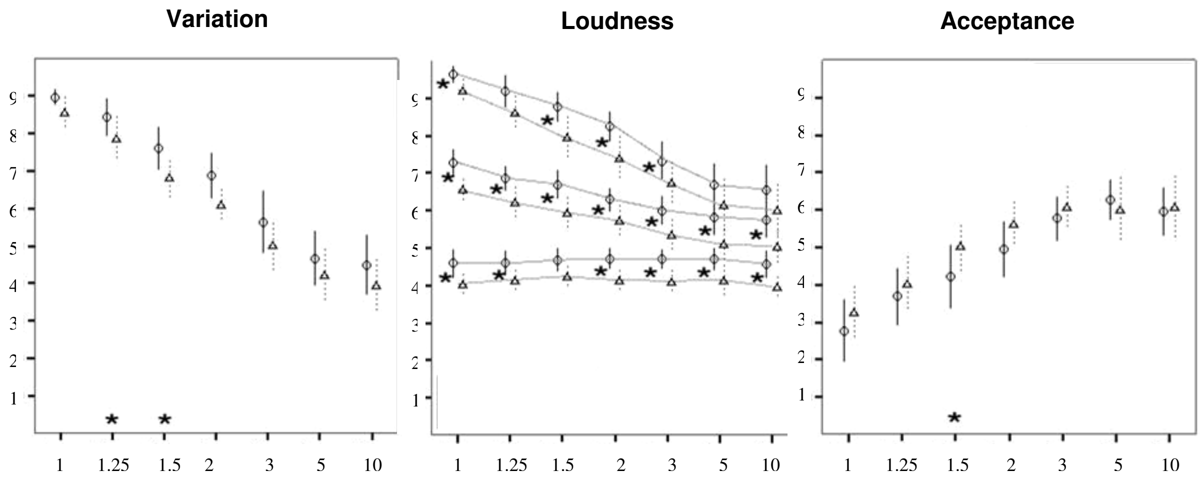

Figure 4.22. Comparison of ratings made for the *Speakers & party*-noise (Δ) and *Football match* (○) signals. (∗) indicate a significant difference between ratings at the given compression ratio ($p < 0.05$). Vertical lines indicate the 95 %-confidence interval for the given mean.

4.4.2 The significance of differences between signal-spectra.

The question is what causes the difference in ratings seen among some of the input signals.

If one compares the long term spectra of signal (1) *Dantale & party*-noise, and signal (2) *Dantale & car*-noise, they are fairly similar (fig. 4.2). The input RMS-levels for speech are the same in all segments in these two signals. And in both signals the input RMS-level for the noise is 5 dB below that of the speech. But the noise spectra in the two signals are different. In signal (1) the party-noise has almost equal energy in the 1/3-octave bands from 100 Hz to 10 kHz, whereas the car-noise has most energy in the 100 Hz region and less energy at higher frequencies compared to the party noise. Also from a psychoacoustic perspective, the party noise is more fluctuating and has a persistent and probably also annoying character, compared to the car cabin noise which is more static in nature. Due to the pronounced energy at the higher frequencies, the party noise will interfere more with the speech signal, making it somewhat harder for the listener to comprehend what is being said. In addition, signals containing party noise may be more tiring for subjects to listen to in the long run.

It might be that subjects listen to the noise in the gaps between speech sentences, and then use the noise as a reference when judging the overall signal. Therefore the *Dantale & car*-noise is judged as being more acceptable (or less annoying) at the lower compression ratios compared to the same signal with party noise. Thus, in this case the RMS-level for speech, and the fact that RMS-levels for the noises are equal in both signals, is a poor predictor for the perceived loudness and acceptance of the two signals.

Regarding signal (3) *Speakers & party*-noise, changes in the shape of the speech spectrum at the raised and loud vocal efforts (see fig. 4.3-4.5) did not result in significantly different ratings, compared to the reproduced speech levels in the *Dantale & party*-noise. But, although not significant, signal (3) did receive lower ratings of variation and loudness and higher ratings of acceptance compared to the *Dantale & party*-noise, within same compression ratios. This was especially the case for the lower ratios. The issue whether reproduced and natural speech at higher levels is perceived differently by listeners needs further investigation. The party noise spectrum was the same in both signal (1) and (3). This may also contribute to the lack of a significant difference in ratings between the two signals, because the speech spectra might have been perceived to be equally dominating and annoying in both signals.

Signal (4), the *Football match*, had RMS-levels in each of the four segments equal to the speech levels in signals (1), (2) and (3). In the 3rd segment, this signal had more energy than all other signals above 5 kHz, and this may have contributed to the significantly higher ratings of variation and loudness. Also, the context of this signal (recording of a sports event with people shouting) may also have influenced some of the subjects to produce higher ratings – especially those not interested in sports or those easily annoyed by loud sounds. The psychological aspects of tolerance for loud sounds are discussed in chapter 6.

4.4.3 Relationship among scales at the "tolerable" compression ratio.

In figure 4.23, the mean ratings of acceptance across compression ratios are shown for the four input signals in the experiment. It can be seen how the acceptance ratings for the *Dantale & car*-noise signal always received higher ratings of acceptance, whereas the *Dantale & party*-noise and the *Football match* received the lowest ratings.

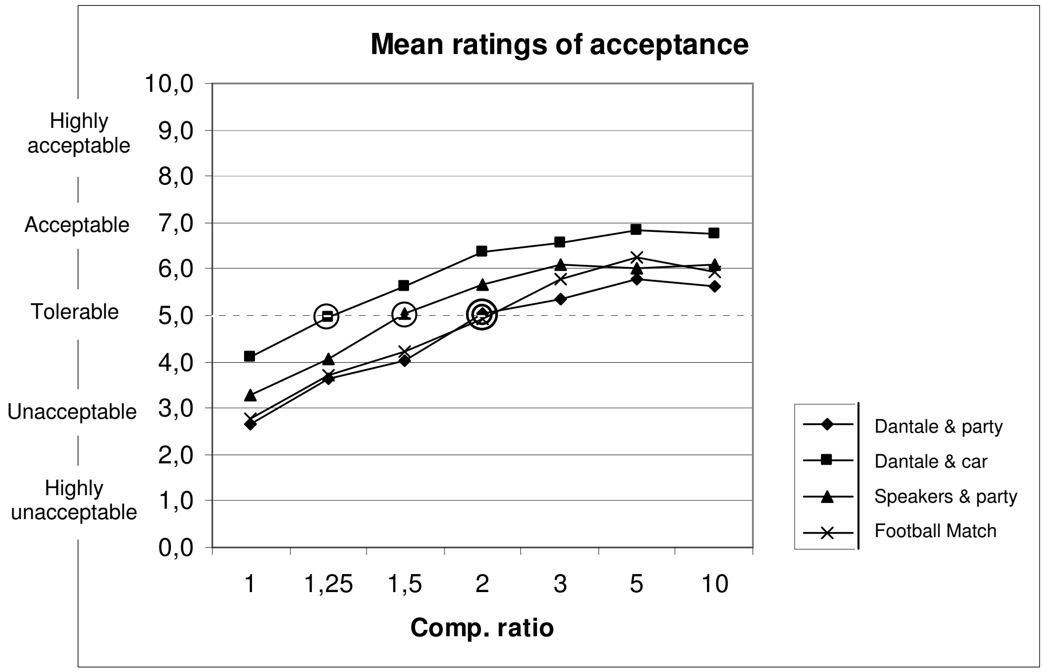

Figure 4.23. Comparison of mean ratings made on the acceptance scale in the four input signals, as a function of compression ratio.

The difference in acceptance ratings among signals indicate, that applying the same amount of gain for all type of signals may not be the optimal choice for a fitting rationale. In this experiment, the gains applied to the loudest part of the signals (i.e., the 2nd, 3rd and 4th segments) were dependent on the compression ratio.

One may draw on the information in figure 4.23 to make a criterion for finding the acceptable compression-ratio in the case of these four signals. The dotted line in the figure divides the acceptance scale into an upper and lower part. The line is set at the "tolerable"-category on the scale (5.0), leaving the categories "unacceptable" and "highly unacceptable" below the line, and "acceptable" and "highly acceptable" above the line. If the fitting-objective is to provide the listener with level variation (in this case, for loud sounds) but still keeping the listening comfort at an acceptable level, then a compression ratio yielding a rating of "tolerable" may be desirable. As indicated in figure 4.23, the ratio yielding a rating of "tolerable" was **1.25:1** for the *Dantale & car*-noise, **1.5:1** for the *Speakers & party*-noise and **2:1** for the *Dantale & party*-noise as well as for the *Football match*.

The decision of dividing the acceptance scale at the "tolerable"-category is an arbitrary one. It seems reasonable though to allow a certain gain for high input-levels as long as the listener does not perceive this as being "unacceptable", or in the range close to this category (i.e., below the dotted line in fig. 4.23). Listening to loud sounds will always be more demanding upon the listener. Even normal-hearing listeners have been found to prefer lower than normal calculated loudness for loud signals (Smeds, 2004a). Also, by choosing an acceptance rating of "tolerable" as the lowest borderline, the listener can be provided with more gain for loud sounds than would be the case if a higher criterion on the scale was chosen.

In figure 4.23, it should be noted that subjects tended not to use the highest category on the scale, named "highly acceptable". Some subjects noted that they did not see the relevance of this category, or that they could not distinguish between *acceptable* and *highly acceptable*. The same issue goes for the lower part of the scale, where subjects tended not to use the category "highly unacceptable" even though the loudness at the lower compression ratios in some cases were judged to be *uncomfortably loud*. Due to this issue, it might have been better to use other adjectives for this scale, or make a finer scale with the lowest category being *unacceptable* and the top most being *acceptable*.

In figure 4.24 (a-c) the relationship between the mean ratings on the acceptance scale and the ratings made on the variation and loudness scales, are shown for each signal. For the loudness scale, the mean ratings of the third (and loudest) segment in the signal is shown, as subjects indicated this segment was decisive for their rating on the acceptance-scale.

In a given signal, the compression ratio yielding an acceptance rating of "tolerable" may be compared to the corresponding ratings on the variation and loudness scales. This provides information about the highest degree of loudness and perceived level variation that could be obtained, without compromising listening comfort. In figure 4.24, these comparisons are marked with a circle.

Table 4.7 summarises the tolerable compression ratios for each signal, based on the criterion used in figure 4.23 and the corresponding ratings of *variation* and *loudness*. As noted earlier, the ratios yielding a rating of "tolerable" are in the range of 1.25:1 and 2:1. The *Dantale & Party noise* and the *Football match* having the highest ratio, and the Dantale & car noise the lowest ratio. The ratings of variation are in the range from 6 to 7 giving a perceived variation in the range of "Midway" to "Large". The ratings of loudness are in the range from 7.7 to 8.2, giving a perceived loudness in the range from "loud" to "very loud".

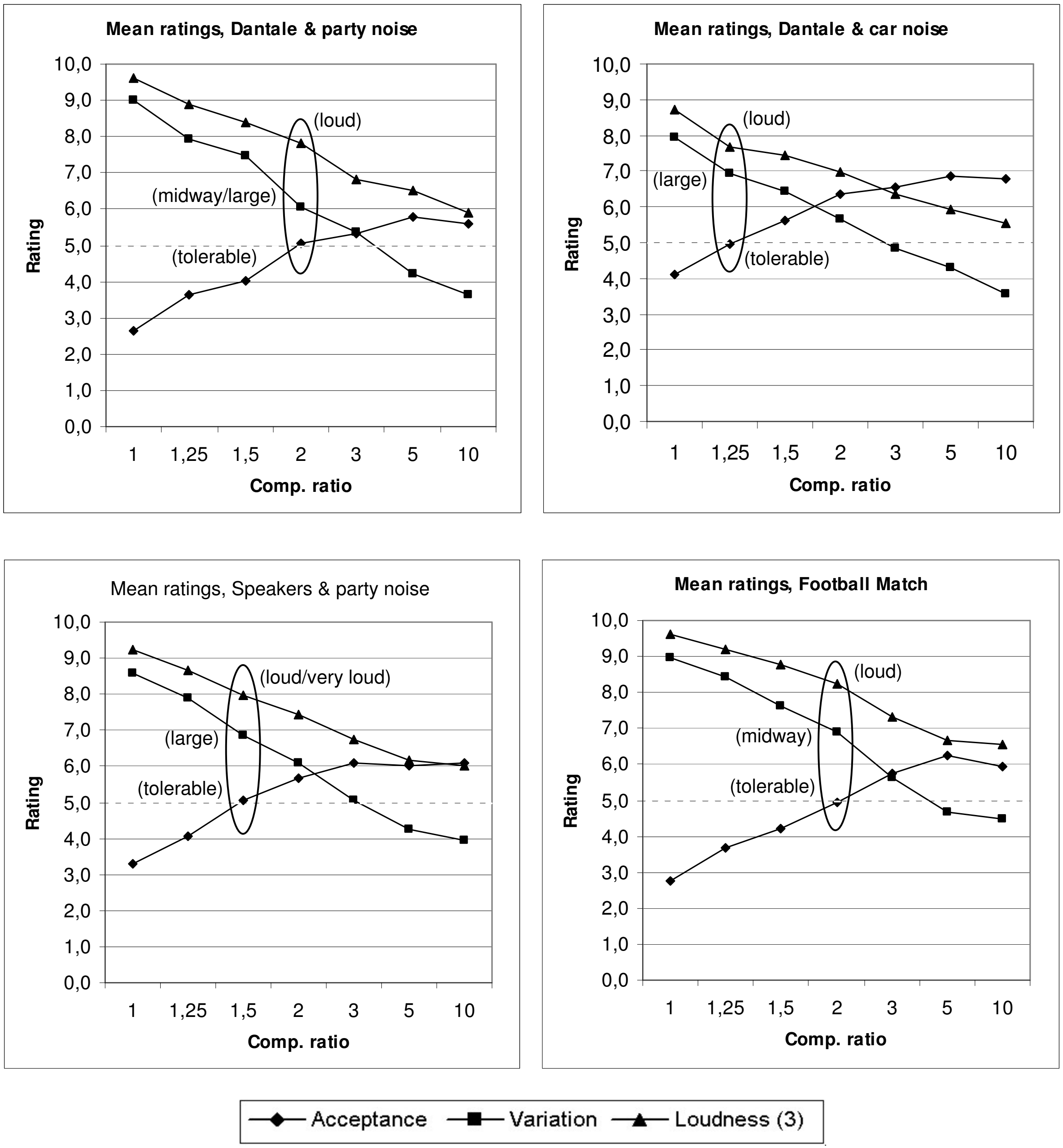

Figure 4.24(a-c). Comparison of ratings on the three scales; Variation, Loudness (3rd segment) and Acceptance for each of the four signals in the experiment. Ratings made at the ratio yielding an acceptance-rating of "tolerable" are encircled (see the text for details).

Table 4.7. The tolerable compression ratio for each signal based, on the division of the "acceptance"-scale, and the corresponding mean ratings on the variation-scale and loudness-scale (3rd segment).

	Dantale & party noise	*Dantale & car noise*	*Speakers & party noise*	*Football match*
Optimal CR	**2:1**	**1.25:1**	**1.5:1**	**2:1**
Variation	6.0 (Midway/Large)	7.0 (Large)	6.9 (Large)	6.9 (Large)
Loudness (3rd segment)	7.8 (Loud)	7.7 (Loud)	8.0 (loud/ V. loud)	8.2 (Loud/V. loud)
Acceptance	5.1 (Tolerable)	5.0 (Tolerable)	5.0 (Tolerable)	4.9 (Tolerable)

The comparison in figure 4.24(a-c) shows that for the signals and RMS-input levels in this experiment, listeners generally were able to accept a loudness having the category “Loud”, and even approaching “Very loud” in the case of the *Speakers and party*-noise signal. The corresponding sensation of variation between segments was approaching the category “large” in most cases. The perceived variation was at no point approaching the category “Small variation”, at the ratio yielding a “Tolerable”-rating on the acceptance scale.

Thus for the loud input signals in this experiment, the objective of presenting the listener with some of the level variation in the input signal was fulfilled. The RMS-levels in this experiment were arbitrarily chosen to illustrate loud sounds. The rating on the three scales would have been different if other signals and other level-intervals had been used. But there may exist a certain relationship between the level, spectrum and noise components of the input signal and the degree of gain for loud sounds that can be accepted by listeners.

4.4.5 Considerations on preferred loudness and signal duration

Using the criteria in figure 4.23, the ratings of loudness (3rd segment) were within the range of “Loud” to “Very loud” in all signals, at the ratio yielding an acceptance-rating of “Tolerable”. The input RMS-level of this segment was 82 dB SPL. In the Smeds (2004a, 2004b) and Neuman et al (1995b) studies, subjects preferred the loudness for similar input-levels to be around the moderate-loudness category.

Although the results in this study cannot be directly compared to the earlier studies, it seems that listeners may accept higher sound levels (and a higher loudness) when listening to signals with level variation, compared to a situation were signals are presented at a fixed level. Continuing along this path, it may be that the optimal signal processing for loud signals should depend on the duration of the given signal level. That is, loud sounds with a longer duration may need to be given lesser gain, compared to loud sounds with shorter duration occurring as part of a level-fluctuating signal. These issues need further investigation, and a suggestion for a field study is given in chapter 6.

4.5 Conclusion of experiment #1

In the present experiment, an alternative approach was used to investigate the perception of level-fluctuations in loud speech and noise signals. Four input signals with a built-in level variation were compressed with seven different ratios in a simulated slow-acting hearing aid. The compression ratios spanned over the range from 1:1 to 10:1, yielding both higher and lower applied gain for the input-levels used, than would normally be seen in a commercial hearing aid. The focus of this study was not on loudness normalisation as such, but on the preferred or acceptable loudness for loud sounds.

Three different research questions were investigated.

Firstly, it was found that hearing-impaired listeners are able to perceive differences in level variation, when loud signals with built-in level-fluctuation are processed with different compression ratios. In both the variation scale and loudness scales (2nd and 3rd segments), a gradual decline in mean ratings was seen with increasing compression ratio. An interesting finding was that the rating of acceptance increased with increasing compression ratio, reaching a rating of “acceptable” at ratios from 3:1 to 10:1. This is in contradiction with preference for a lower ratio (< 3:1) found in literature.

Secondly, it was found that spectral differences among signals with equal RMS-levels, do influence the hearing-impaired listeners’ perceptions of loudness and acceptance when the signals are processed with the same settings. Signals differing in the noise-type and spectral characteristics of the noise, were rated significantly different. Dantale in broadband party noise received the highest ratings of loudness and the lowest ratings of acceptance. In contrast to this, there was no difference in ratings between reproduced speech and natural speech, spoken at different vocal efforts. Apart from spectral differences, psychological factors related to the context of the sound may also have influenced the ratings.

Finally, a rating of “tolerable” was chosen on the acceptance scale as the criterion for the lowest allowable compression ratio. With this criterion, the corresponding loudness-levels were higher in this study, compared to earlier studies where the high fixed presentation levels were used. The shorter duration of the high levels in this experiment seemed to increase listeners' tolerance for loud sounds. Both issues mentioned above, indicate the need of a fitting-algorithm that regulates gain for loud sounds, depending on their spectral (and temporal) characteristics as well as their duration.

In the present study, a slow regulating compressor with an attack-time of 100 ms and release-time of 5000 ms was used. Such a combination of time constants is usually not seen in commercial hearing aids. The relatively long time constants were chosen in order to compare with the study by Smeds (2004a) and to reduce the number of parameters for investigation.

Several studies have shown that the sound quality, speech intelligibility and noisiness are influenced by the time constants in combination with the chosen compression ratio (e.g., Neuman et al, 1998). In addition, the effectiveness of the compressor on signals with short duration (like the syllables in speech) will depend on the responsiveness of the compressor – long release-times resulting in less compression of fast fluctuations in the input signal.

There is a need for investigating the influence of time-constants, also for loud input signals to the hearing aid. This was done in a second experiment, using two different speech signals in the presence of noise. This experiment is presented in the following chapter 5 «««.

5. The effects of compression ratio and release-time on loud speech and noise signals, processed by a simulated non-linear hearing aid (experiment #2)

5.1 Introduction and research question

In earlier studies investigating preferred listening levels for soft and loud sounds, a preference has been found for placing such sounds closer to the most comfortable loudness level (Neuman et al, 1995; Smeds et al, 2004a, 2004b). A consequence of this finding would be that a compression strategy with a rather high compression ratio should be chosen, at least for high input levels. In this way, the level range of loud sounds would be narrowed in, before being presented to the hearing aid user.

But the sound processing of level differences not only depends on the compression ratio, but also on the dynamic characteristics of the compressor. Apart from the compression ratio, this includes the attack- and release-times, the lower threshold of the compressor and the number of channels in the hearing aid. In combination with the type of input signal, the setting of these parameters will affect the effective compression ratio, the overall output level and the short-term spectral and temporal level differences of the processed signal (Kuk & Ludvigsen, 1999). This again could affect the listeners' impressions of loudness as well as sound quality and speech intelligibility.

In experiment #1 (described in chapter 4) a slow-acting compressor with an attack time of 100 ms and release time of 5000 ms was used. In such a slow system, only the long term level differences will be regulated by the hearing aid, leaving the short term variations (like syllables in speech) untouched. Thus, the specified ratio of compression will only be obtained for very slow modulations in the input signal. For faster modulations, the effective ratio will decrease depending on the attack- and release-times of the compressor (Verschuure et al. 1996). When implemented in a multichannel hearing aid, such slow systems have been found to increase listening comfort in hearing-impaired listeners, yielding better scores of sound quality and speech intelligibility (e.g., Hansen, 2002).

On the contrary, a compressor with short time constants will be more effective in compressing the faster modulations in the signal. This is the basis for the concept of "syllabic compression", where attack- and release-times are in the range of 10-50 ms (Dillon, 1996). In such a system the soft syllables in speech will receive more gain compared to syllables with higher intensity. Theoretically, this could increase speech intelligibility by decreasing the vowel-to-consonant ratio (that is, increasing audibility for soft speech sounds). But negative effects related to the manipulation of the natural relationship between soft and loud syllables in speech, and to the introduction of amplitude fluctuations which tend to fuse speech and background noise together, has also been put forward (Plomp, 1994; Stone & Moore, 2003).

Many commercial hearing aids make use of fast-acting compression schemes. But some manufactures employ a slow-acting scheme, often in combination with a fast-acting compressor, which regulates the gain in case of abrupt level changes in the input signal. In addition, most digital hearing aids make use of several compression channels with different compression settings that may work independently of each other.

As a continuation of earlier studies, it is also relevant to investigate the effects of the dynamic aspects of compression, when the input signal is either loud speech or noise. In the present study the combined effects of three different release times and six compression ratios were investigated, using the same test-setup as in experiment #1. Two different speech and noise signals presented at 75 dB SPL were used as input to the experimental compressor. One signal contained speech at a loud vocal effort at a poor signal-to-noise ratio. The second signal contained speech at a normal vocal effort, but reproduced at high level with a favourable signal-to-noise ratio. The processed versions of the two signals were presented in the free field, and subjects listened via hearing aids fitted to NAL-R for the individual hearing loss.

The specific purpose of the experiment was to investigate how differences in speech spectra and signal-to-noise ratio would influence the subjective impression of loudness, speech intelligibility, noise nuisance and overall acceptance of the processed signals. Knowledge gained from this experiment might be useful for the setting of time-constants and compression ratios in an adaptive hearing aid that regulates its gain depending on the input-spectrum and signal duration.

In summary, this experiment attempts to answer the following research questions:

- Do differences in speech spectra and signal-to-noise ratios between signals cause a significant difference in listeners' perceptions of the signals, when processed with the same compression settings?
- Which combination of compression ratio and release-time provides the "best impression" of speech intelligibility and user acceptance (and the lowest noise nuisance), in the two signals?

5.2 Method

5.2.1 Input signals

Two different input signals containing speech and party noise were prepared in the sound-editing software (Adobe Audition, version 1.0):

(1) Male speaker at loud vocal effort & party noise (0 dB SNR).
(2) Male speaker at normal vocal effort & party noise (+15 dB SNR).

The spectral shapes of the two vocal efforts were in accordance with the spectra specified in the ANSI-S3.5 standard (1997) (see also below).

An illustration of the speech and noise-waveforms in one signal is shown in figure 5.1. Each signal had a duration of 50 seconds. The speech and noise levels were kept the same throughout the signal – that is, there was no overall level variation in the signal as in experiment #1. The two signals were to be sent to the experimental compressor, at a speech input level corresponding to 75 dB SPL. This is the RMS-level specified for loud vocal effort in the ANSI-S3.5 standard (1997). The choice of vocal efforts and signal-to-noise ratios was made to simulate two realistic situations, where the hearing aid user is listening to loud speech in background noise.

Signal (1), containing speech at a loud vocal effort, was made to simulate a situation where the listener attends a party with a high noise level. In this situation, he or she enters a conversation with a male talker, who needs to raise his voice to be heard above the noise in the room. The signal-to-noise ratio of 0 dB was chosen to be an appropriate simulation of a noisy situation, where two speakers move at a certain distance of each other to increase speech understanding (Ross, 1992).

Signal (2) containing speech at a normal vocal effort, was made to simulate a situation where the hearing aid user is listening to a radio or TV-set at a high volume. In the given broadcast, a speaker talks at normal vocal effort. The noise in the background simulates some reality-sound in that feature, which is attenuated 15 dB relative to the speech-level, in order to maintain a favourable signal-to-noise ratio.

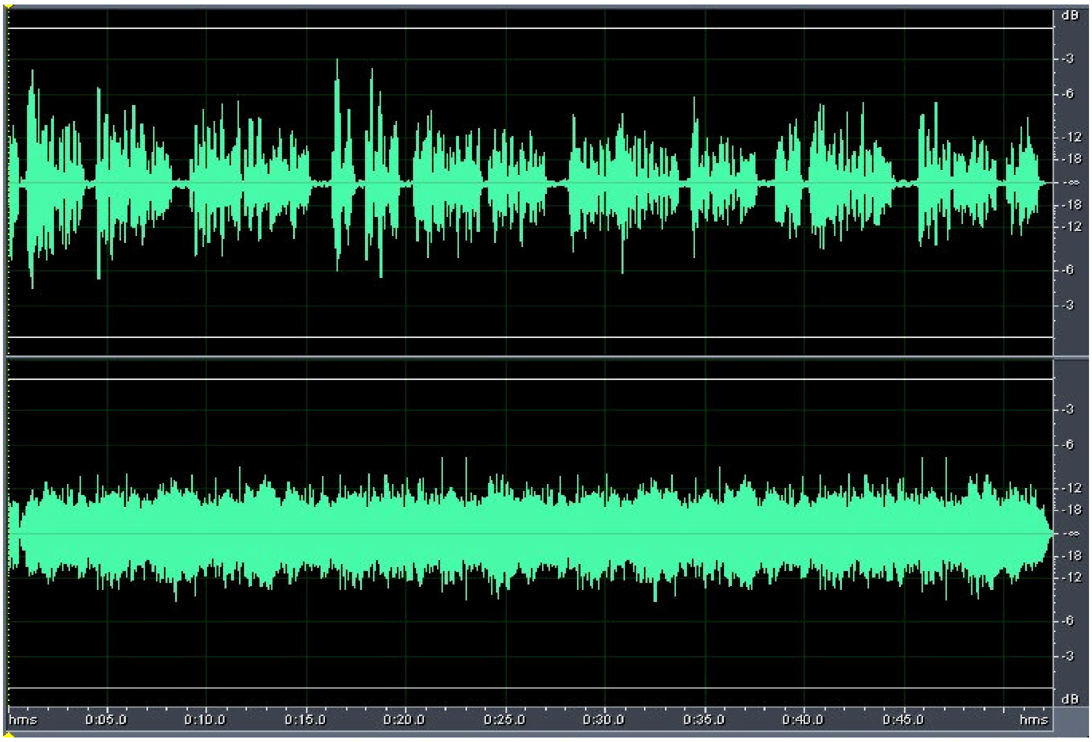

Figure 5.1. Example of waveforms in one of the input signals to the compressor. Speech and noise parts in signal (1): *Loud speech & party*-noise (0 dB SNR).

The speech parts for signals (1) and (2) were taken from a recording of a male speaker, reading from a text about a whale-expedition (Widex, personal communication). In this recording, the voice level was controlled to match RMS-levels for normal and loud vocal efforts, as specified in the ANSI S3.5 standard (ANSI, 1997). This was done by presenting noise over headphones at various levels, thereby forcing the speaker to raise his vocal effort in order to keep monitoring his own voice.

The noise part used in both signals was a recording of a party situation, taken from a compact disc containing environmental sound examples (Widex, 1999).

Long-term spectra for the two speech signals used in signals (1) and (2) are shown in figure 5.2. The equal RMS levels of the two speech signals are kept in the figure.

Compared to speech at a normal vocal effort, the spectrum for the loud vocal effort has more energy in the region from 400 to 1500 Hz (average of 4.6 dB) and less energy below 300 Hz (average of 14 dB, excluding the region at 150 Hz). Thus, the spectrum for loud speech shows the shift in slope that would be expected from the change in vocal effort.

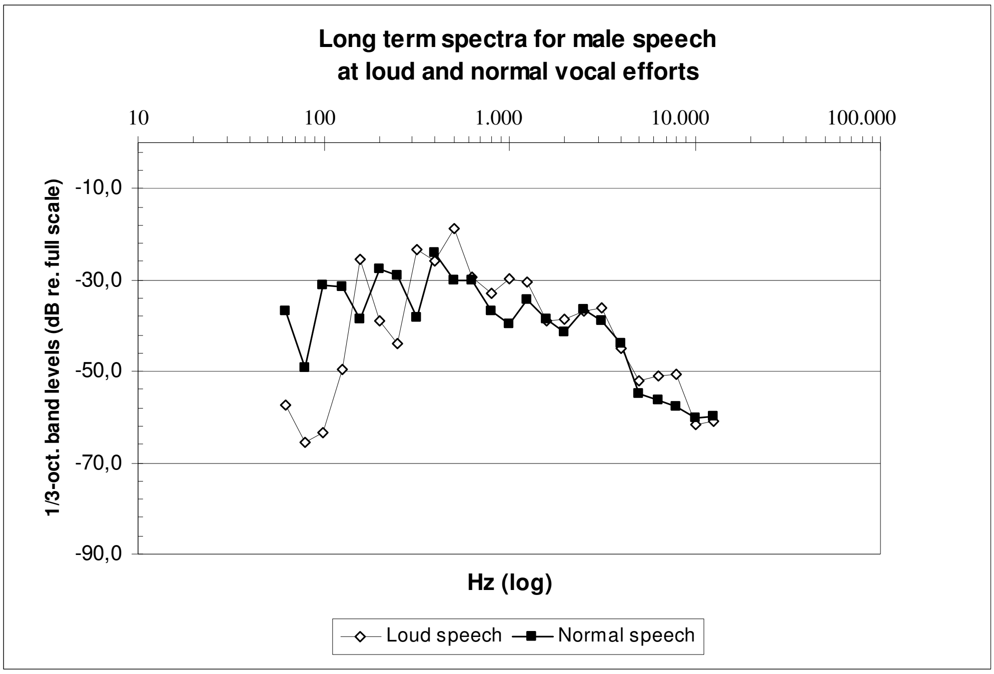

Figure 5.2. Long-term spectra (1/3-octave levels) for male speech at loud vocal effort used in signal (1), and for male speech at normal vocal effort used in signal (2). The equal RMS levels of the two speech signals are kept in the figure.

Figure 5.3 shows the individual spectra for the speech and noise in signal (1). The relative RMS level difference of 0 dB between the speech and noise are kept in the figure. Note that the noise signal has more energy below 300 Hz (average of 13.1 dB) and above 4000 Hz (average of 9.6 dB), compared to the speech signal. In this case, the audibility of the unprocessed speech signal is expected to be greatly compromised due to masking from the noise.

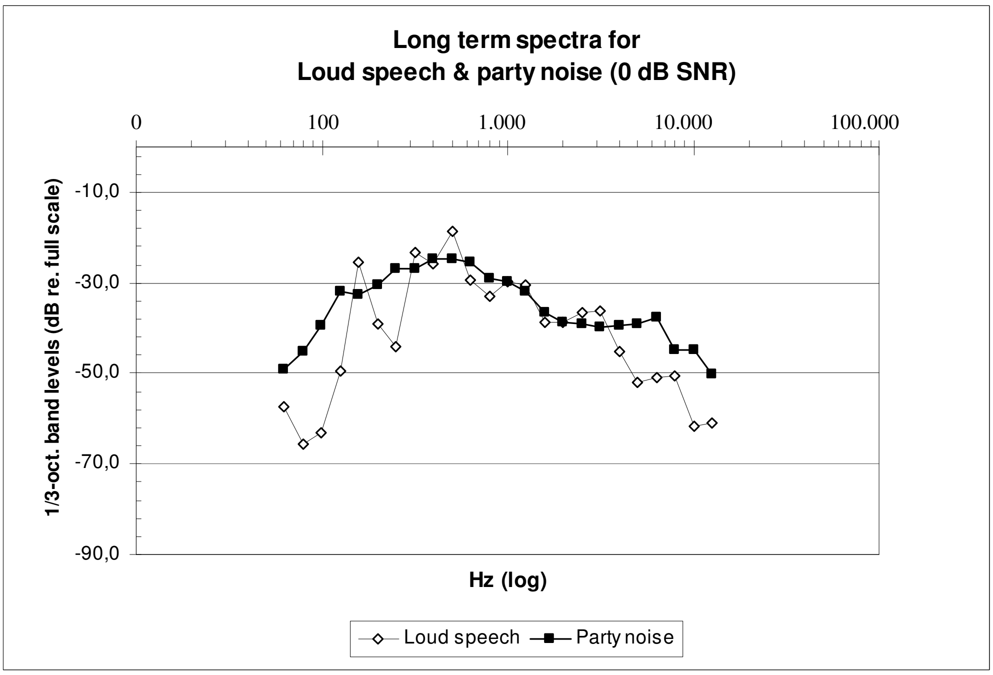

Figure 5.3. Long-term spectra (1/3-octave levels) for male speech at loud vocal effort and for the party noise used in signal (1). The SNR between speech and noise signals is 0 dB.

The individual long term spectra for the speech and noise in signal (2) are shown in figure 5.4. Due to the positive signal-to-noise ratio of 15 dB, the speech-spectrum has more energy than the noise signal at most frequencies (average of 12.7 dB up to 5000 Hz). In this case, the intelligibility of the unprocessed speech is expected to be high.

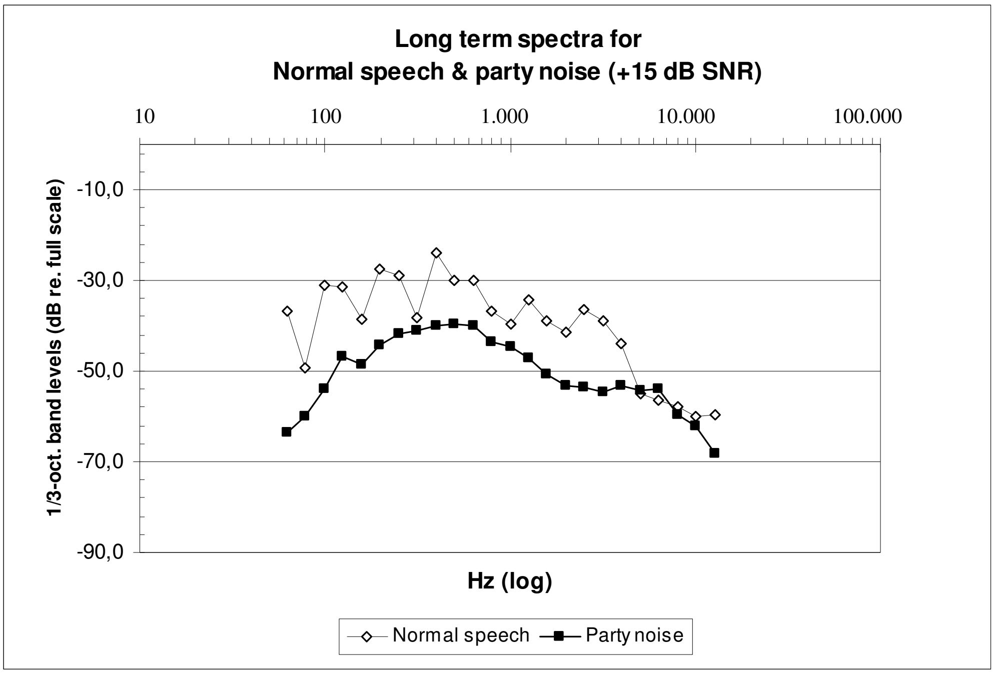

Figure 5.4. Long-term spectra (1/3-octave levels) for male speech at normal vocal effort and for the party noise used in signal (2). The SNR between speech and noise signals is +15 dB.

Long term spectra for both signals (1) and (2), with speech and noise mixed together, are shown in figure 5.5. As part of the intention to present signals at equal RMS-levels, the two spectra are overlapping although the vocal efforts and noise levels differ in the two signals.

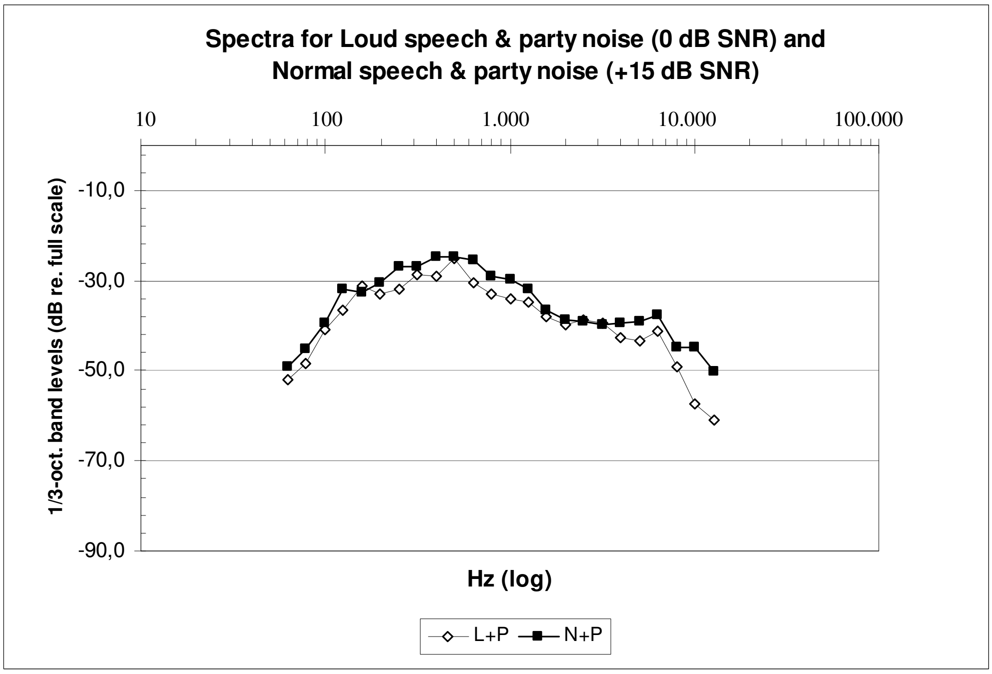

Figure 5.5. Long-term spectra (1/3-octave levels) for signals (1) and (2), with speech and noise mixed.

5.2.2 Compression of signals

The two input signals were compressed offline in the experimental compressor also used for experiment #1. This compressor has three independent compression channels and was implemented as a Simulink-model in MATLAB by Carsten Paludan-Müller (personal communication, Nov. 1993). A detailed description of the compressor can be found in subsection 4.2.2.

The two input signals were compressed with six different compression ratios: 1:1 (linear condition), 1.5:1, 2:1, 3:1, 5:1 and 10:1 and three different release-times of 40, 400 and 4000 ms. The attack-time was always held fixed at 10 ms. The chosen compression ratios were the same ratios used in experiment #1, except that the 1.25:1 ratio was left out for this experiment. The three release-times were chosen to reflect a fast syllabic compressor, a slow compressor and a very slow compressor. Release-times in the range from 40 to 4000 ms are also seen in typical commercial hearing aids.

The input levels for speech in the two test-signals were adjusted to 75 dB SPL RMS-level (i.e. the level specified for loud vocal effort in ANSI-S3.5). The input level was calibrated by sending only the speech from each signal (containing no pauses) through the model, and adjusting the input-gain to a level corresponding to 75 dB SPL (or -21 dB re. full scale). The anchor-points (i.e., the levels in each channel receiving the same gain in dB, regardless of the compression ratio) were adjusted to the RMS-levels measured at the second measurement-pin, while sending normal speech (with pauses removed) through the model at an input RMS-level of 62 dB SPL (see appendix 9.2).

This simulated a commercial hearing aid with a handle that varies the degree of gain (or compression ratio) for high input levels - while always keeping the same gain for a normal speech input of 62 dB SPL. The gain applied to the two test signals would vary depending on the given compression ratio – that is, gain would be reduced with increasing ratio.

The broadband static input-output characteristics of the compressor are shown in figure 5.6. The overall RMS input level of the two input signals (75 dB SPL), and for the normal speech signal used for adjusting the anchor-points (62 dB SPL), are encircled on the abscissa.

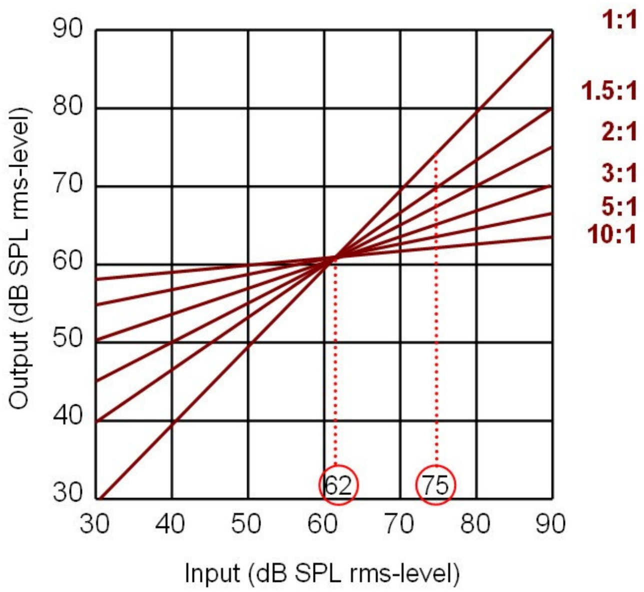

Figure 5.6. Illustration showing the broadband input-output characteristics of the experimental compressor. The RMS input levels for speech in the two test signals was 75 dB SPL. The anchor-points in each channel were adjusted relative to a normal speech input at 62 dB SPL RMS-level.

5.2.2.1 Compensation for the influence of release-times

When generating the compressed signals, using the combination of compression ratios and time-constants mentioned above, the gain applied by the compressor would also be influenced by the release-time. With the short release-time of 40 ms, the applied gain (and the output level from the compressor) would be reduced, compared to the condition with a long release-time of 4000 ms. This influence would become greater with increasing compression ratio.

To compensate for the influence of release time, the output RMS-levels for a normal speech input of 62 dB SPL were measured in all combinations of ratios and release-times. Then, in all cases, the level was raised to the output level of the uncompressed condition (1:1). This would simulate a hearing aid that always keeps the same gain for a normal speech input, regardless of changes in compression ratio and time-constants. The dB-values used for compensating the change in output level for each combination of ratio and release-time are shown in table 5.1. It can be seen that the need for adjustments was greatest in conditions with the shortest release-time and high degrees of compression.

Table 5.1. The dB-values used at each compression ratio to raise the output-level from the compressor, in order to compensate for the level-reduction caused by changes in the release time.

	10/40	**10/400**	**10/4000**
1.5:1	2.32 dB	0.98 dB	0.31 dB
2:1	3.09 dB	1.65 dB	0.44 dB
3:1	3.60 dB	1.96 dB	0.58 dB
5:1	3.76 dB	2.17 dB	0.59 dB
10:1	3.77 dB	2.26 dB	0.61 dB

5.2.2.2 Characteristics of output signals from the compressor.

The interaction between release-time and compression ratio is expected to influence the dynamic characteristics of the output signals from the compressor. In figure 5.7 (a-f), long term spectra of the processed versions of the two input signals are shown for each of the three release-times.

For signal (1) *Loud speech & party*-noise (left column), the highest output response is seen in the 1:1-condition and the lowest in the 10:1-condtion. The difference between responses is greatest at the short release-time of 40 ms. Here the RMS-level at the 10:1 condition is 16 dB lower than in the linear condition. A smaller reduction of 13.3 dB occurs with the release time of 4000 ms. For signal (2) *Normal speech & party*-noise (right column), the difference between output responses is not as big. Still the measured RMS-level for the 10:1 condition is 11 dB below that of the linear condition. Approximately the same reduction in RMS-level is seen at all three release times.

The output responses for signal (2) are generally lower than for signal (1). This is assumed to be caused by the noise being 15 dB below the speech in that signal, compared to signal (1) where speech and noise have same RMS-levels. Thus in signal (1), the broadband party noise will bring more energy to the signal over the whole frequency range and will fill out pauses in the speech - yielding a higher RMS level in the processed versions of this signal. On the contrary, in signal (2) the noise does not have the same influence in the speech pauses, and does not contribute as much to the overall energy of the signal - yielding a lower measured RMS-level for the compressed versions of this signal.

The greater signal-to-noise ratio in the *Normal speech & party*-noise, may partly explain the similarity between output-responses for this signal. The experimental compressor has an infinitely low compression threshold, as depicted in figure 5.6. The anchor-points were set according to the RMS input-level for normal speech (62 dB SPL). The RMS input level of signal (2) was 75 dB SPL for speech and therefore the RMS-level for noise in that signal would be 60 dB SPL. This means that the noise in signal (2) would receive more gain with increasing compression ratio. And therefore the influence of the noise on the measured RMS-level (and the output response) would be greater at higher compression ratios.

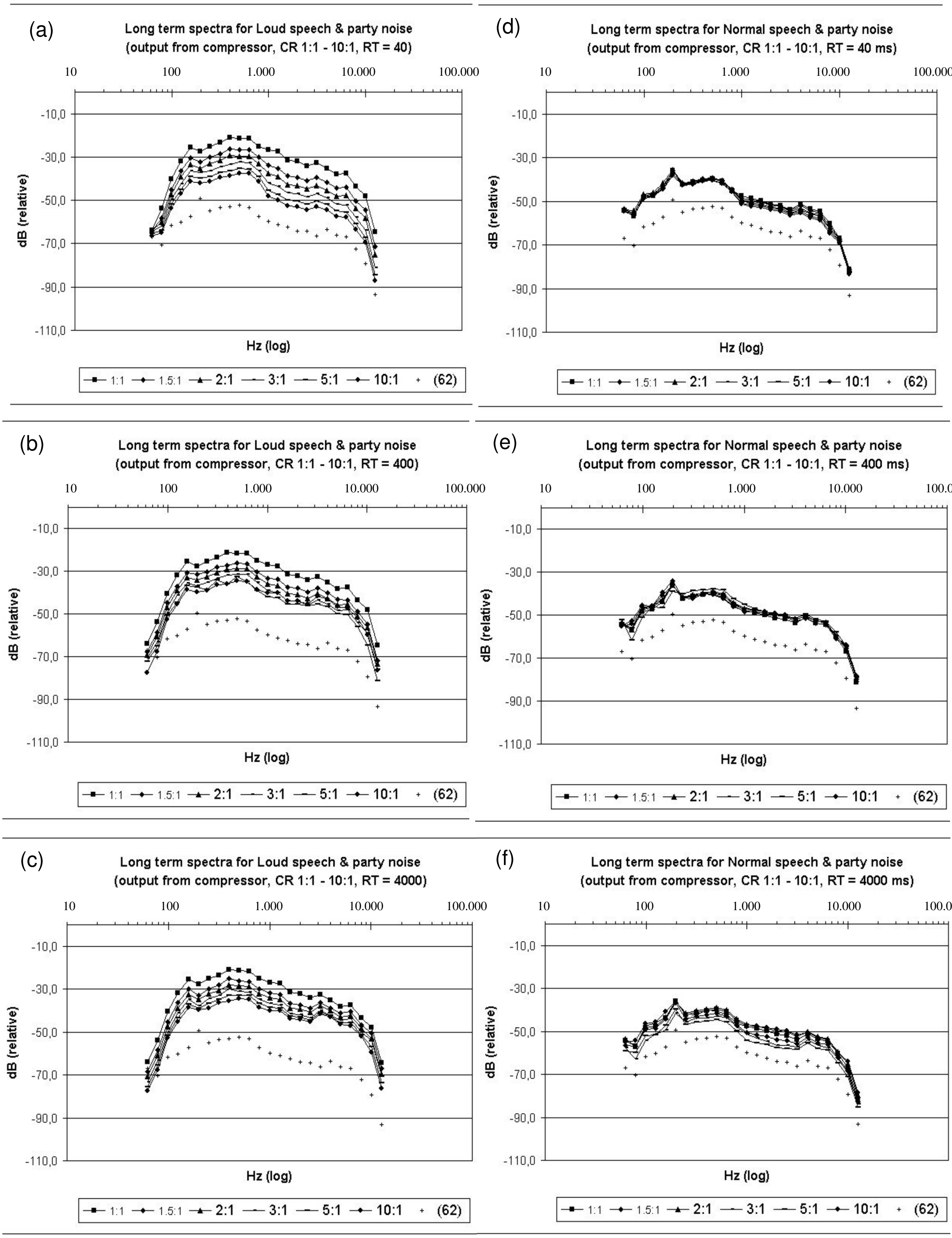

Figure 5.7(a-f). Long-term spectra (1/3-octave levels) for the compressed input signals, shown for RT's of 40, 400 and 4000 ms. Spectra for the *Loud speech & party noise* are shown to the left and for the *Normal speech & party noise* to the right. "+"-signs indicate the spectrum for normal speech (62 dB SPL input level), sent through the compressor in the linear condition.

In figure 5.8 (a-f) the minimum, average and maximum RMS-levels[2] as a function of compression ratio, are shown for each of the three release-times. In both signals, the average RMS-levels are reduced with increasing compression ratio. At a release time of 40 ms, the range between the min and max levels diminishes in both signals with increasing compression ratio. This is most obvious in the *Normal speech & party*-noise, where the reduction in range is 22.4 dB between the 1:1 and 10:1 conditions, compared to only 13 dB in the *Loud speech & party*-noise. At the 400 and 4000 ms release times, the range does not change to the same extent, which would be expected as the slow compression applied to the signal only affects the overall output level of the signals.

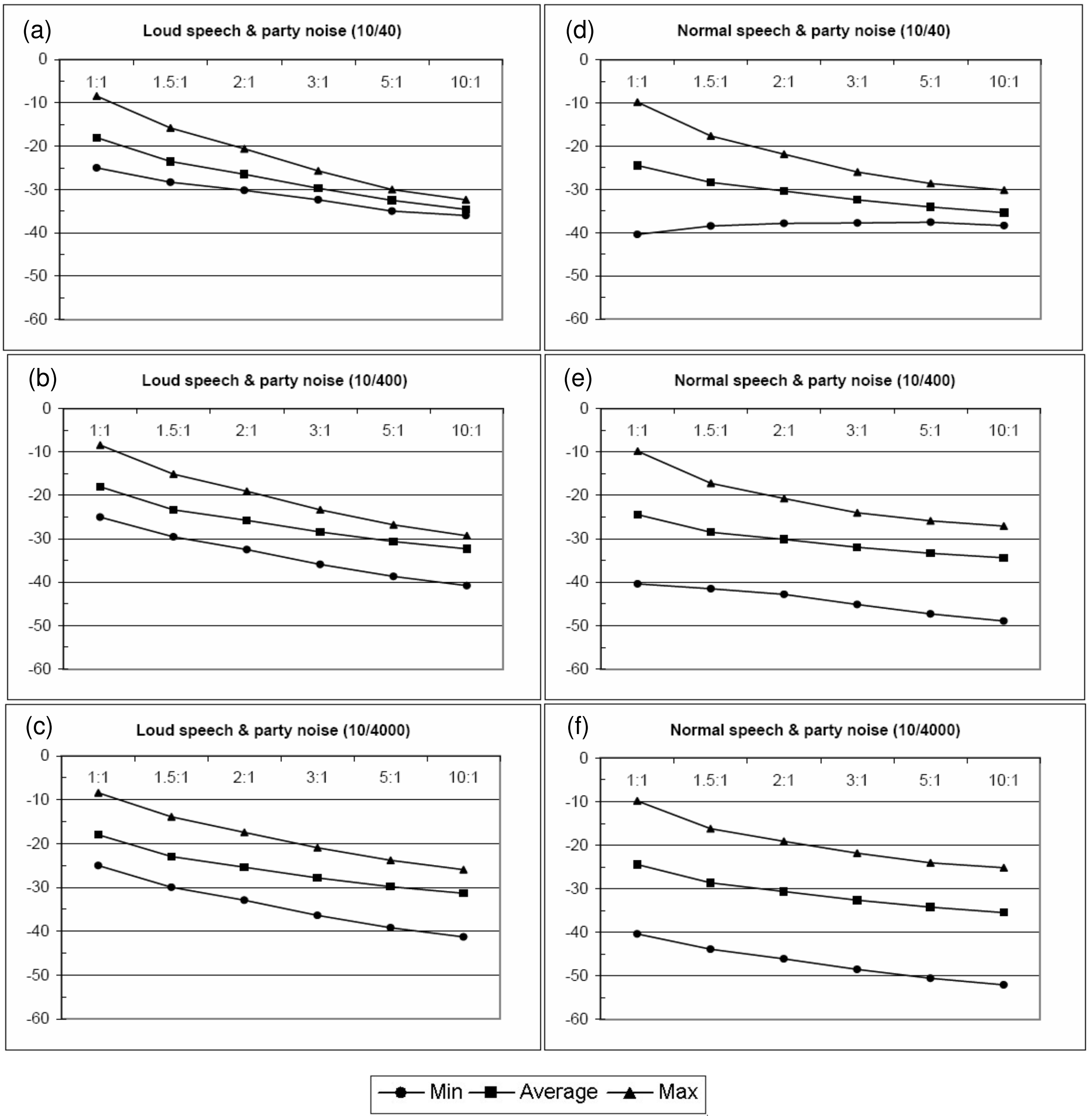

Figure 5.8(a-f). Minimum, average and maximum RMS-levels as a function of compression ratio, for each of the three release-times. Levels for signal (1) are shown in the left and for signal (2) to the right.

2 The root mean square levels for the input and test signals were measured in the sound editing software used for generating the test signals (Adobe Audition, 2003). Measurements were done by selecting all parts of the signal (apart from the initial fade in and final fade out) using a window-width of 50 ms. From this selection the software calculates three RMS-levels, the minimum, the maximum and the average level. The minimum and maximum RMS-levels are the lowest and highest window-values found in the chosen selection. The average RMS-level is the average of all of the sums of the minimum and maximum values from the window sections in the selection (S. Garnett, personal communication, May 17th, 2006).

The two signals differ in regard to the minimum RMS-levels which are generally lower in the *Normal speech & party*-noise signal. This is believed to be caused by the lower noise level in that signal, and the fact that the speaker talks slower at this normal vocal effort - with pauses of longer duration between sentences.

Sound examples of the two input signals, as well as the compressed test signals can be found on the audio-CD in appendix 9.1.

5.2.3 Test setup for listening experiment #2

5.2.3.1 Presentation of test signals

As in experiment #1, test signals were presented to listeners in the anechoic chamber. Signals were played back from a laptop computer, using the sound-editing program. The signal was routed through an external USB-soundcard (Creative Estigy) to the loudspeaker amplifier (QUAD 606). For this experiment, a KEF Caprice II loudspeaker was used for presenting the signals (see appendix 9.3). This loudspeaker has a power handling of 100 Watt and a frequency response of +/- 2.5 dB in the range from 68 Hz – 20 kHz.

5.2.3.2 Calibration of presentation levels

Test subjects were seated at a distance of three meters from the loudspeaker. White noise with a bandwidth of 750 Hz, centred at 1 kHz, was used to calibrate the presentation level at the position of the listener. The calibration noise was given a level equal to the RMS-level of a reference signal containing normal speech at 62 dB SPL input level, which had been sent through the compressor in the linear condition (1:1). This was the same signal that was used for setting the anchor point of the compressor.

A Brüel & Kjær sound level meter, type 2240 was used to calibrate the test setup. Before each listening test, the calibration signal was adjusted to 62 dB SPL (+/- 1 dB) at the position of the listener. In this way, normal speech, processed in the linear condition, would be presented at 62 dB SPL at the microphone of hearing aids worn by subjects. The compressed versions of the two signals would be presented at levels above 62 dB SPL, depending on the ratio used for the given signal.

5.2.3.3 Hearing aids worn by test subjects

Similar to experiment #1, all subjects wore binaural BTE-hearing aids (Widex Senso Diva, see appendix 9.4), fitted linearly according to the National Acoustic Laboratories Revised (NAL-R) fitting procedure (Byrne & Dillon, 1986). Hearing aids were fitted with custom ear moulds, with 1.2 mm ventilation channels. The purpose of the hearing aids was to make the compressed signals audible, placing normal speech at 62 dB SPL at the most comfortable level of the listener. The combination of the compressed signals presented from the loudspeaker and the linear gain in the hearing aids, should simulate a non-linear hearing aid.

NAL-R targets for insertion gain were calculated from individual hearing thresholds at 500, 1000, 2000 and 4000 Hz. Targets were inserted in the fine tuning screen of the software used for fitting the Senso Diva Hearing aid (Widex Compass, version 3.4.1). A further description of the NAL-R procedure and settings made in the hearing aids can be found in subsection 4.4.3. The fitting data for all subjects, together with test box-measurements of gain- and output, can be found in appendix 9.5.

5.2.3.4 Test-subjects

Seven hearing-impaired listeners with moderately sloping losses participated in the study. They were three males and four females ranging from 60 – 85 years of age, with a mean age of 74.6 years. Subjects were tested to have normal middle ear function. Threshold configurations for right and left ears of all subjects are shown in figure 5.9. The seven subjects were all experienced hearing aid users and participated also in experiment #1 of this project.

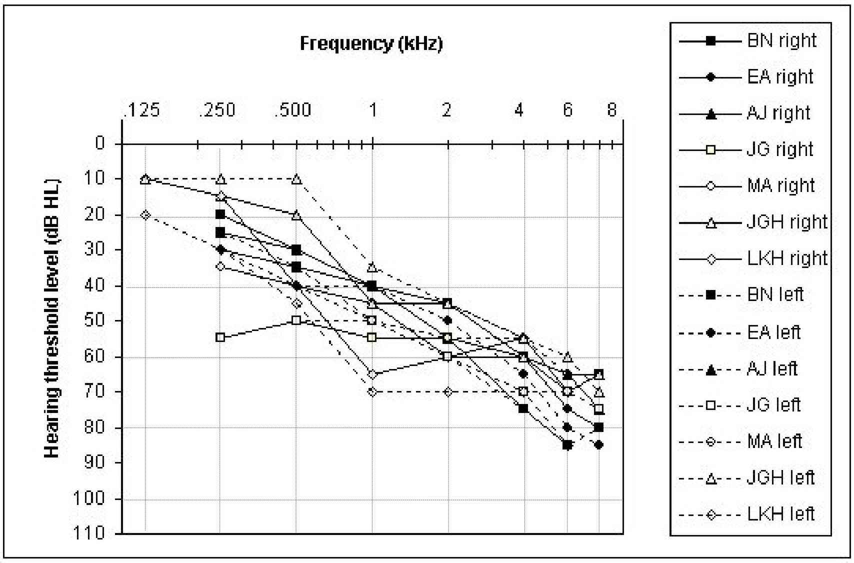

Figure 5.9. Hearing thresholds for right and left ears of the seven subjects in experiment #2.

5.2.4 Procedure for listening experiment #2

The sixteen compressed versions of the two input signals were presented three times in randomised order to each test subject, giving a total of 48 trials. Subjects rated each trial on four categorical scales (discussed below). Before the actual test began, six randomly chosen signals were presented for training purposes.

Testing was done in two sessions of approximately one hour (on two separate days). The trials belonging to the *Loud speech & party*-noise were presented in the first session and trials belonging to the *Normal speech & party*-noise in the second session. The choice of dividing the two signals into two sessions was made to avoid subjects becoming confused if both signals were presented within the same session.

The second session also included three presentations of normal speech at 62 dB SPL RMS-level, sent through the compressor-model in the linear condition. This was done to validate whether the anchor point of the compressor was adjusted appropriately for a 62 dB SPL speech input (see fig. 5.6), and if the hearing aids succeeded in amplifying normal speech to the most comfortable loudness level. Table 5.2 provides an overview of the two test sessions.

Table 5.2. Overview of the two test sessions in experiment #2.

Session 1	Session 2
Loud speech & party noise *(0 dB SNR)*	*Normal speech & party noise* *(+15 dB SNR)*
16 compressed versions of signal (1) presented 3 times in random order = 48 trials. (Randomized for each subject). Six of the signals presented for training purposes, before the test	16 compressed versions of signal (2) + *Normal speech* at 62 dB SPL, presented 3 times in random order = 51 trials. (Randomized for each subject). Six of the signals presented for training purposes, before the test

5.2.4.1 Subject instruction and rating on categorical scales

In the beginning of the session, test subjects were presented with the original input signal. For the *Loud speech & party noise* they were instructed to imagine themselves being at a party, having a conversation with a person standing in front of them. For the *Normal speech & party noise* they were asked to imagine themselves listening to a speaker's voice, coming from a radio or TV-set at a high volume setting. They were then told that the versions of the original signal presented during the test had been processed through different hearing aids. This processing led some of the signals to be softer than the original, and also the noise in signals might be modified in different ways.

During the actual test, each compressed signal was presented in loops, while the listener made their ratings on four psychometric scales with a pencil (figure 5.10 and appendix 9.9). The construction of these scales was similar to the ones used in experiment #1, but the categories were different.

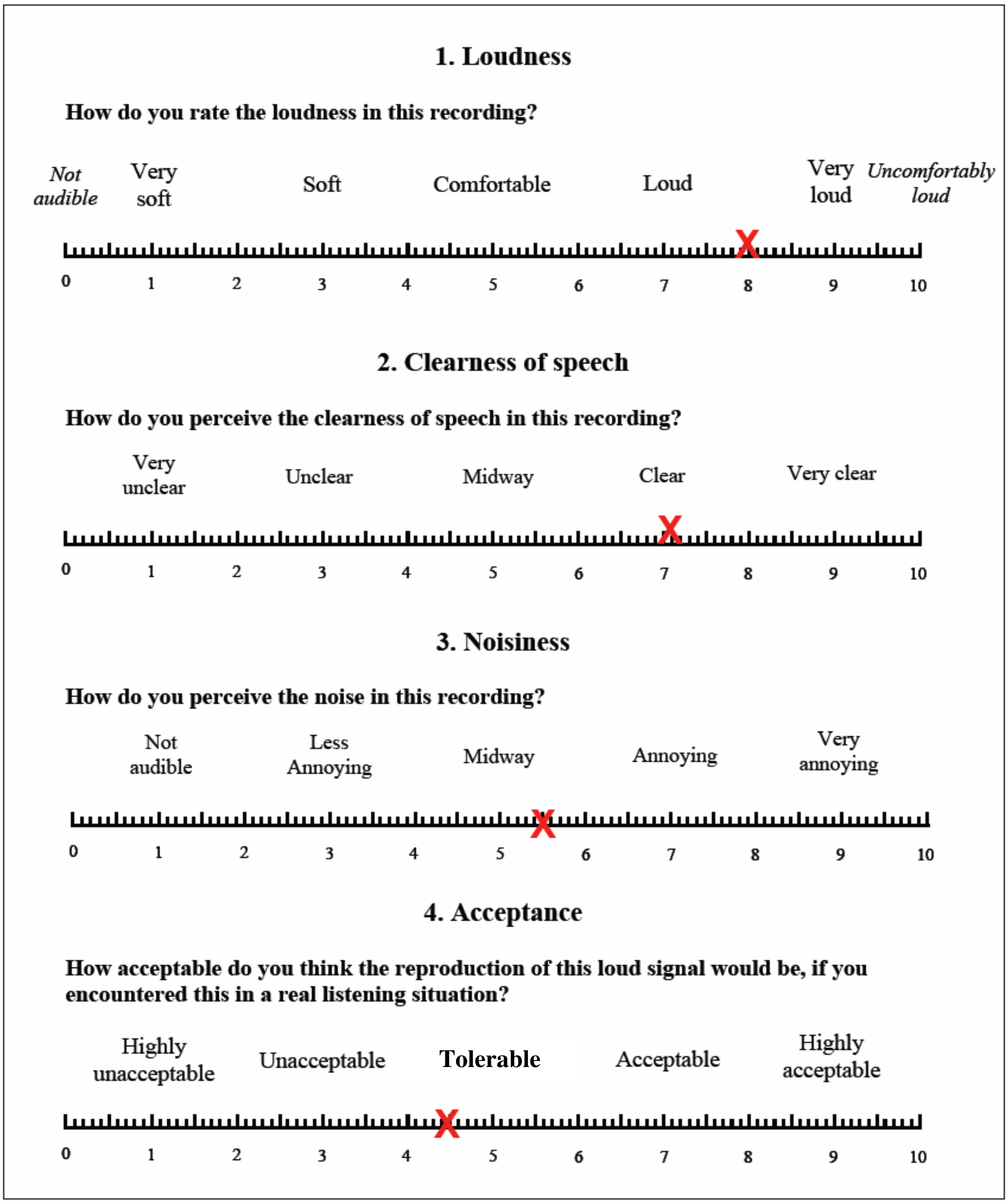

Figure 5.10. The four categorical scales used for the subjective rating in experiment #2 (English translation).

On the first scale (**Loudness**), listeners were asked to mark the loudness they perceived for the overall signal (both speech and noise) in that recording. This scale contained the main categories; “Not audible”, “Very soft”, “Comfortable”, “Loud”, “Very loud” and “Uncomfortably loud”.

On the second scale (**Clearness of speech**), listeners were asked to mark how clear they perceived the speech in the given recording. They were instructed that the question about *clearness* could be rephrased as “how clearly do you understand what is being said?” This scale contained the main categories; “Very unclear”, “Unclear”, “Midway”, “Clear”, “Very clear”.

On the third scale (**Noisiness**), listeners were asked to mark how annoying they perceived the noise on that recording. This scale contained the main categories; “Not audible”, “Less annoying”, “Midway”, “Annoying”, “Very annoying”.

Finally on the fourth scale (**Acceptance**), listeners were to mark how acceptable the reproduction of the original signal would be, if encountered in a real situation. This scale was also used in the experiment #1. The main categories of this scale were “Highly unacceptable”, “Unacceptable”, “Tolerable”, “Acceptable” and “Highly acceptable”. Subjects were asked to judge how acceptable this hearing aid setting would be, if they needed to listen to the sound for 5-10 minutes.

The adjectives used for the four scales were inspired by Neuman et al (1998) who also used categorical scales, originally proposed by Gabrielsson et al. (1979, 1990). They used scales to assess subjective impressions of clarity, background noise and preference of signals processed with different compression characteristics.

For this experiment, the chosen scales were intended to provide information about the effect of compression ratio and release-time on speech understanding and noise nuisance, when the hearing aid places a loud input signal in the upper part of the listener’s auditory range. The interaction between compression-settings and the two different signal-to-noise ratios used in signals (1) and (2) was also of interest. Finally information on the perceived loudness and overall acceptance of the processed signals would provide guidance for formulating a fitting rule, which would aim at presenting loud sounds such that they are perceived as being louder than moderate - but still comfortable and providing the listener with the best possible speech understanding.

In the training part of each session, subjects were presented with six test signals that had been compressed with different ratios and release-times. During the presentations, subjects tried to mark on the four scales and were able to ask questions.

All ratings made during the actual listening test (that is, 16x3 in the first session and 17x3 in the second session – excluding ratings made in the training-part) were collected and entered into a data spreadsheet for further analysis.

5.3 Results

In figure 5.11(a-h), mean ratings of loudness, clearness of speech, noisiness and acceptance are shown as a function of release time and compression ratio. Means for the *Loud speech & party*-noise (signal 1) are shown in the left column and for the *Normal speech & party*-noise (signal 2) in the right column. Error-bars indicate the 95 % confidence interval for the given mean. In each figure, the mean ratings made in the linear condition (1:1) are shown only for the 40 ms release time, as this condition was tested only once (1x3) in each signal.

On each of the four scales, a mixed model analysis of variance was carried out, using statistical software (SPSS, version 11.5). The fixed effects for the dependent variable Rating in each scale are shown in tables 5.3 to 5.10. The full data output from the ANOVA made for each scale together with model verification, can be found in appendix 9.11.

A post hoc comparison with Bonferroni-correction was made to find significant differences in ratings ($p < 0.05$) of signals being processed with the same release time. The horizontal lines above columns in fig. 5.11 indicate that means under the line do not differ significantly from each other ($p < 0.05$). In some cases, the point-of-view when comparing means is indicated by a black dot, showing that the mean at this compression ratio does not differ significantly from the other means under the line.

(1) *Loud speech & party*-noise (0 db SNR)

Table 5.3 Loudness-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6	684,069	,000
COMP	15	312,000	70,800	,000
TRIAL	2	312,000	5,544	,004

Table 5.4 Speech-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6	196,153	,000
COMP	15	312	15,537	,000
TRIAL	2	312	,425	,654

Table 5.5 Noise-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6	434,885	,000
COMP	15	312	5,929	,000
TRIAL	2	312	6,537	,002

Table 5.6 Acceptance-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6	847,084	,000
COMP	15	312	4,384	,000
TRIAL	2	312	5,806	,003

a = Dependent Variable: RATING.

(2) *Normal speech & party*-noise (+15 dB SNR)

Table 5.7 Loudness, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6,012	481,292	,000
COMP	16	315,013	73,861	,000
TRIAL	2	315,503	,438	,646

Table 5.8 Speech-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	5,966	699,770	,000
COMP	16	314,967	52,600	,000
TRIAL	2	315,594	,998	,370

Table 5.9 Noise-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	5,987	162,266	,000
COMP	16	314,987	44,482	,000
TRIAL	2	315,284	,166	,847

Table 5.10 Acceptance-scale, tests of Fixed Effects(a)

Source	Numerator df	Denominator df	F	Sig.
Intercept	1	5,938	443,951	,000
COMP	16	314,939	22,195	,000
TRIAL	2	315,758	,650	,522

(1) Loud speech & party-noise (0 db SNR) **(2) Normal speech & party-noise (+15 dB SNR)**

Figure 5.11(a-h) Mean ratings of loudness, clearness of speech, noisiness and acceptance as a function of release time and compression ratio. Means for signal (1) are shown to the left, and for signal (2) to the right. Error-bars indicate the 95 % confidence interval for the given mean. Horizontal lines indicate no significant difference between means under the line.

The ANOVA showed a significant effect of the compression setting (i.e., the combination of compression ratio and release-time) on all scales and in both signals. An effect of trial was also seen in signal (1), in the loudness scale (table 5.3), the noisiness scale (table 5.5) and the acceptance scale (table 5.6). This effect was not intended, and may be caused by insufficient training in some subjects. A closer inspection of the data showed a slight reduction in the mean scores of loudness and noisiness, from the first to the third trial. In the acceptance data, a slight increase was noted. The variation in ratings were generally small (within 5 tick marks on the scale) and do not question the validity of the overall means. But the variance is expected to be greater in these cases.

In figure 5.11 (a & d), the mean ratings of **Loudness** in both signals diminish with increasing compression ratio. This tendency is the same for all release times, but the span of ratings is greatest at RT = 40 ms. At this release time, the highest mean ratings are close to “Very loud” and the lowest are close to “Soft”. At release times of 400 and 4000 ms a narrower range of ratings is seen, especially in the *Normal speech & party*-noise signal. At these two release times, signals are being rated to be less loud at the lower ratios and louder at the higher ratios, compared to the release time of 40 ms. For both signals (1) and (2), ratings seems to “even out” as release time increases. At a ratio of 10:1 and RT = 4000 ms, both signals receive ratings close to “Comfortable” loudness.

Ratings of the **Clearness of Speech** show different patterns in the two signals (fig. 5.11, b & d). In the *Loud speech & party*-noise signal, the means never exceed the 6th mark, dividing the categories “Midway” and “Clear”. The opposite is found with the *Normal speech & party*-noise, where the highest mean ratings are close to the 8th mark, in-between “Clear” and “Very clear”. In both signals, mean ratings diminish with increasing ratio, with the lowest mean being close to “Unclear” in signal (1) at CR = 10:1 and RT = 40 ms. Again, ratings seem to even out at the longer release times. In signal (1) there are no significant differences between means at 4000 ms, whereas in signal (2) only the mean at 10:1 is significantly lower than other ratings at this release time.

The ratings of **Noisiness** also differ between signals (fig. 5.11, c & e). In the *Loud speech & party*-noise ratings are in-between the categories “Midway” and “Distracting” at all ratios and release times (except at 1:1 where the rating is even higher). For this signal there is also a tendency to reduced ratings with increasing release time, but generally there are no significant difference between large groups of means (and not at all for the 400 ms release time). For the *Normal speech & party*-noise, ratings made at the 40 ms and 400 ms release times increase with increasing ratio, reaching the category “Distracting” at ratios of 10:1. But also here there is a tendency for the ratings to even out, and at the 4000 ms release time there are no significant differences between ratings. In this condition, all ratings are close to the “Not distracting” category.

Finally in figure 5.11 (d & f), the mean ratings of **Acceptance** hardly differ across conditions in the *Loud speech & party*-noise signal. All means lie between the categories “Unacceptable” and “Tolerable”. There is a tendency of diminishing ratings at the 40 ms release time and increasing ratings at the 4000 ms, whereas ratings stay the same at 400 ms. This is also supported by the statistical analysis. The *Normal speech & party*-noise generally receives higher ratings at the lower ratios than signal (1). At higher ratios the ratings diminish, especially at the 40 ms release time. At 400 ms the means become more even, and at 4000 ms they are significantly similar and approaches the “Acceptable”-category.

Two-paired t-tests were also carried out to compare mean ratings in the two signals, processed with the same compression ratio and release time. Significance levels for each comparison are shown in table 5.11. A star indicates a significant difference ($p < 0.05$) in mean ratings between signal (1) *Loud speech & party*-noise, and signal (2) *Normal speech & party*-noise. The signal receiving the highest rating on the given scale is indicated in parentheses.

Table 5.11. T-test comparison of means obtained with same compression ratios and release times in signals (1) and (2). Asterisks indicate significant differences ($p < 0.05$) in ratings between the two signals. The signal receiving the highest rating is indicated in parentheses. (S1) = *Loud speech & party*-noise, (S2) = *Normal speech & party*-noise.

	Loudness			Clearness of speech			Noisiness			Acceptance		
	40	**400**	**4000**	**40**	**400**	**4000**	**40**	**400**	**4000**	**40**	**400**	**4000**
1:1	.049* **(S1)**			.000* **(S2)**			.000* **(S1)**			.001* **(S2)**		
1.5:1	.038* **(S1)**	.001* **(S1)**	.000* **(S1)**	.000* **(S2)**	.000* **(S2)**	.000* **(S2)**	.000* **(S1)**	.000* **(S1)**	.000* **(S1)**	.000* **(S2)**	.000* **(S2)**	.000* **(S2)**
2:1	.155	.068	.000* **(S1)**	.002* **(S2)**	.000* **(S2)**	.000* **(S2)**	.011* **(S1)**	.002* **(S1)**	.000* **(S1)**	.000* **(S2)**	.001* **(S2)**	.000* **(S2)**
3:1	.093	.483	.014* **(S1)**	.001* **(S2)**	.001* **(S2)**	.000* **(S2)**	.009* **(S1)**	.014* **(S1)**	.000* **(S1)**	.017* **(S2)**	.000* **(S2)**	.000* **(S2)**
5:1	.938	.271	.541	.013* **(S2)**	.000* **(S2)**	.000* **(S2)**	.081	.272	.000* **(S1)**	.929	.011* **(S2)**	.000* **(S2)**
10:1	.616	.674	.058	.080	.058	.001* **(S2)**	.004* **(S2)**	.253	.000* **(S1)**	.657	.801	.000* **(S2)**

The *Loud speech & party*-noise received significantly higher ratings of *loudness* and *noisiness*, at the lower compression ratios (1:1 – 3:1) in combination with all three release times. For the noisiness-scale, this signal was also rated to be significantly noisier at the 5:1 and 10:1 ratios in combination with the 4000 ms release time. In contrast, the *Normal speech & party*-noise received significantly higher ratings of *speech clearness* and *acceptance*, in nearly all combinations of compression ratio and release time.

In all four scales, the general trend was that signals were not rated significantly different in combinations of higher compression ratios and shorter release times. One exception is in the noisiness scale, where the *Normal speech & party*-noise was rated as being significantly noisier in the combination of a 10:1 ratio and 40 ms release time.

5.4 Discussion of experiment #2

The present experiment investigated the combined effect of compression ratio and release time on listeners' perceptions of loud input signals. Two speech and noise signals, differing in speech spectra and signal-to-noise ratio were processed in a three channel compressor, with sixteen combinations of compression ratio and release time. In both signals, the input level to the compressor was 75 dB SPL (speech level) and the anchor-points of the compressor were set according to an overall input level of 62 dB SPL.

The difference seen in ratings of the two signals, seem to be governed in large by the level of the noise. For the loudness-scale, the reduction in ratings seen in both signals is presumably related to the lowering of the output level from the compressor at higher compression ratios (see fig. 5.6) and the influence of the release time on the dynamic range of signals (see fig. 5.8).

In signal (1), *Loud speech & party*-noise (0 dB SNR), the noise seems to "drown" the influence of the release time. This is seen in the noisiness scale, where the distraction from the noise is rated as being high in all conditions - although slightly decreasing with increasing compression ratio. This decrease may be related to the reduction in loudness of the overall signal at the higher ratios. In the Acceptance scale, the *Loud speech & party*-noise receives a rating below “Tolerable” in all conditions. Only at RT = 4000 ms, ratings seem to increase - but only the mean at the 10:1 is significantly higher than the means obtained at lower ratios.

The ratings of speech clearness made for signal (1) may be linked to the loudness, in case of the short release time of 40 ms. But at RT = 400 ms and 4000 ms, the more even ratings close to the “Midway” category seem to be related to the influence of the noise. The generally lower impression of speech clearness in this signal may also influence listeners’ overall acceptances of the signal – in this case receiving a rather low acceptance in all conditions. This relationship was also noted by Preminger & Van Tasell (1995), who found that changes in sound quality can only be measured separately from changes in speech intelligibility when the speech is clearly audible above the noise - i.e., when the signal-to-noise ratio is positive.

In signal (2), *Normal speech & party*-noise (+15 dB SNR), the influence of the release time on the speech and noise signals is more clearly seen. In the ratings of speech clearness, the reduction in means with increasing ratio is most prominent at 40 ms and 400 ms. Even though the speech ratings may be linked to the loudness of the signal, they also seem related to the release time and its influence on the speech signal.

In combinations of a short release time and high compression ratio, the dynamic range of signal (2) becomes narrower (as shown in fig. 5.8). This could mean that the natural intensity-relationship between soft and loud speech components becomes distorted. At the same time, the ratings of noisiness increase with compression ratio, which is the opposite of the pattern in signal (1). This may be due to the increased gain for noise at the lower side of the anchor point in the compressor (fig. 5.6). This makes the noise increasingly audible in the speech pauses and may thereby also affect the intelligibility of the speaker.

At the longest release times of 4000 ms, the noise-level in speech pauses is re-established to the original SNR, and ratings of speech clearness remains high whereas noisiness-means remain at a low level. This is also reflected in the ratings of acceptance where equally high ratings are given in all condition at this release time.

Test subjects in this experiment also stated that difficulties understanding the speech in signal (1) and the distraction from the noise, made this signal tiresome in all situations. In contrast, the better audibility of speech in signal (2) made it easier for them to detect differences between the processed versions of this signal.

The difference in mean ratings between release times is more clearly seen in figure 5.12 (a-g). In these plots, mean ratings made at the three release times are shown as a function of compression ratio. Brackets indicate no significant difference in means ($p > 0.05$) between release times.

From figure 5.12 it is seen how the shortest release time of 40 ms produce the lowest ratings of *loudness* (a, d) and *speech clearness* (b, e). At the higher compression ratios, the means obtained at the three release times generally differ significantly from each other. Highest ratings of *speech clearness* are obtained in both signals at the 4000 ms release time. Ratings of noisiness (c, f) do not differ from each other in signal (1), whereas in signal (2) the longest release time stands out as the one producing the lowest noise nuisance.

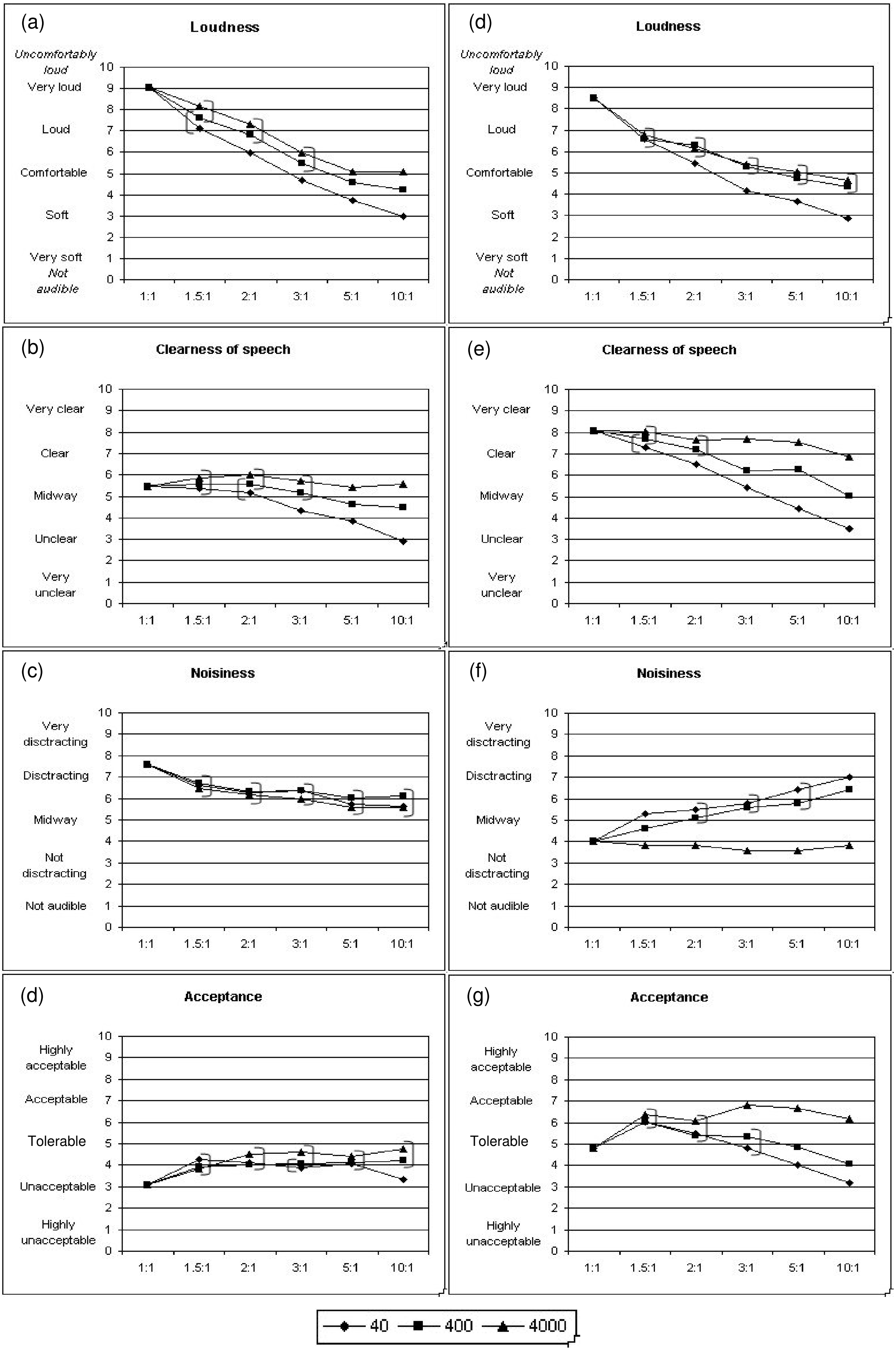

Figure 5.12(a-g). Mean ratings of loudness, clearness of speech, noisiness and acceptance as a function of release time (ms) and compression ratio. Means for signal (1) are shown to the left and for signal (2) to the right. Brackets indicate no significant difference in means (p > 0.05) between release times at the given compression ratio.

The lack of a preference for release time in signal (1) is seen in the acceptance scale (d, g), where all combinations of release time and compression ratio receive the same lower rating. However in signal (2) some significant spreading in the ratings occurs at higher ratios, and the release time of 4000 ms receives the highest ratings in all conditions. At RT = 40 and 400 ms, the rating of acceptance falls close to the “Tolerable”-category when the compression ratio is 3:1, and it falls below this category at ratios of 5:1 and 10:1.

5.4.1 Relationship among scales at the “best” combination of release time and compression ratio

The test signals in this experiment represent two listening situations where the hearing aid user is presented with loud speech and noise. Signal (1) simulates a conversation at a noisy party and in signal (2) the hearing aid user listens to a radio or TV-broadcast at a high volume setting.

If the objective is to present the hearing aid user with the level variations in the environment, the goal should be to present such sounds in the upper part of the listener’s audible range, while still keeping user satisfaction at an acceptable level. The question arise what combination of ratio and release time achieve this goal for the two input signals in this experiment?

One may use the same criterion as in experiment #1, where the acceptance scale is divided into an upper and lower part with the category “Tolerable” as midline (equal to a rating of 5). Due to the nature of the loud sounds, a minimum rating of “Tolerable” would be a reasonable goal, whereas higher ratings falling at the “Acceptable” and “Highly acceptable” categories would be less likely for such signals - even in normal-hearing listeners.

From figure 5.12, the highest mean ratings of acceptance were made at a release time of 4000 ms, although not significant in signal (1) and at the lower ratios in signal (2). Anyhow, it seems clear from a satisfaction-perspective that this release time provides the highest amount of speech clarity and acceptance in both signals. For signal (2), this release time also produces the lowest ratings of noise nuisance.

In figure 5.13, mean ratings on all scales made at the release time of 4000 ms are compared with each other. For the *Loud speech & party*-noise, ratings of speech clearness and noisiness do not change significantly across ratios, and it is difficult to chose a “best ratio” as all the acceptance-means lie below the “Tolerable”-category. A compression ratio of **2:1** seems most appropriate, as the rating of acceptance is closer to “Tolerable” compared to the 1:1 and 1.5:1 ratios.

Also the perceived noisiness is lower at this ratio and the loudness is perceived as being “Loud”, which should be appropriate in relation to the speaker’s loud vocal effort in this signal. Due to the poor signal-to-noise ratio, this signal is generally difficult to handle for the simulated hearing aid. Such signals require additional processing (noise reduction or microphone directionality), to raise speech intelligibility and perceived user comfort.

The mean ratings of acceptance in the *Normal speech & party*-noise are above the “Tolerable”-category at all compression ratios. Therefore the lowest ratio of **1.5:1** would be the best choice for this signal, yielding a loudness of “Loud” and speech being perceived as in-between “Clear” and “Very clear”. Due to the greater SNR in this signal, the perceived noisiness is kept low in all cases with the 4000 ms release time.

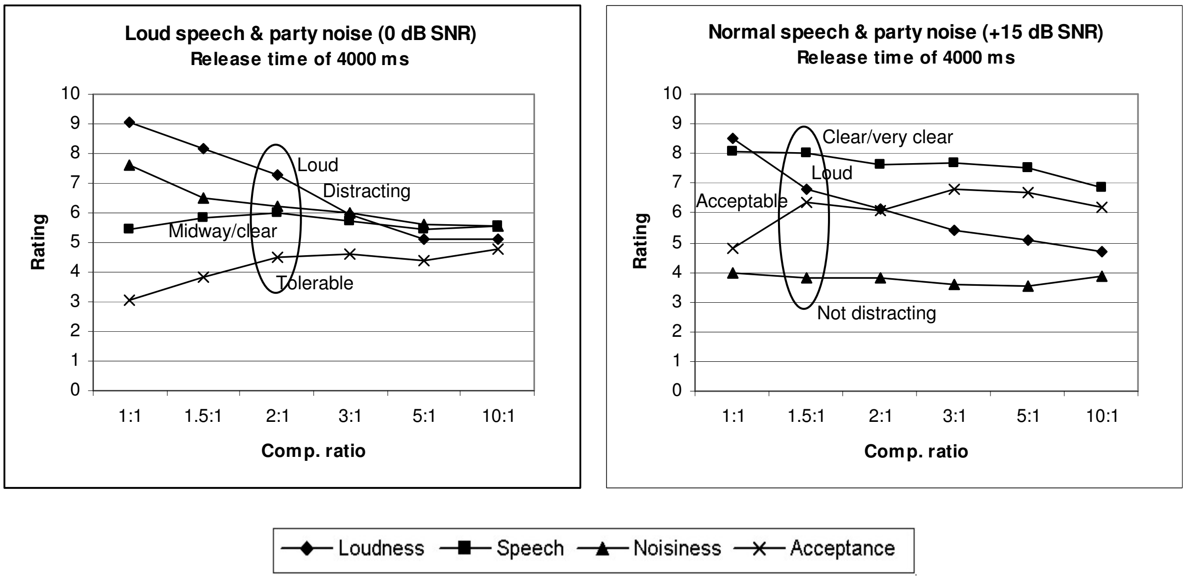

Figure 5.13. Mean ratings at the four categorical scales as a function of compression ratio, made at the release time of 4000 ms. Means for signal (1) are shown to the left and for signal (2) to the right. Mean ratings at the "best" compression ratio (being the ratio where mean ratings of "acceptance" is close to or above the "Tolerable"-category) are encircled in both plots. The category-labels closest to the ratings at the "best" compression ratios are also shown.

5.5 Conclusion of experiment #2

The present study investigated the effect of release times and compression ratio on listeners' impressions of loudness, speech clarity, noisiness and overall acceptance. Two signals containing speech at loud and normal vocal efforts, and party noise at 0 dB and +15 dB SNR, were used as input to the experimental compressor. The signals were sent to the compressor at an RMS-level corresponding to 75 dB SPL (for speech), and with the anchor-points of the compressor adjusted according to a normal speech input at 62 dB SPL RMS-level.

Two different research questions were investigated in this experiment.

Firstly, it was found that difference in signal-to-noise ratio between signals did result in significant differences in ratings on the four scales. The favourable SNR of +15 dB in signal (2) *Normal speech & party*-noise, made the influence of the release time clearer compared to signal (1), *Loud speech & party*-noise, where the poor SNR seemed to even out ratings in all conditions.

Secondly, the most optimal combination of compression ratio and release time appeared to be 2:1 for signal (1) and 1.5:1 for signal (2) – in both cases when using a long release time of 4000 ms. With this combination, the highest degree of speech clarity and lowest possible noisiness was achieved, while still maintaining an acceptance of "Tolerable" and a realistic loudness for the two signals.

The results of this experiment are in accordance with earlier studies that investigated the effects of compression ratio and release time. Arguments against the use of high compression ratios in combination with shorter time constants have been put forward in studies focusing on different attributes of sound quality. Neuman et al (1994, 1995, 1998), using a single channel compressor, found that compression ratio had the greatest impact on subjective ratings of sound quality made by hearing-impaired listeners. And when the compression ratio was 3:1, a short release time of 60 ms gave significantly lower ratings of sound quality, compared to release-times of 200 and 1000 ms. This was especially the case for signals with poor signal-to-noise ratios. Overall, good sound quality was preserved when the compression ratio was below 3:1.

Similarly, Hohmann and Kollmeier (1995) and Boike and Souza (2000), using relatively short release-times of 7 and 70 ms, also found that speech quality-ratings decreased with increasing compression ratio. Also, Hansen (2002), using a multi-channel compressor, found that subjects preferred longer release-times (4000 ms) in combination with low compression thresholds (20 dB SPL), over short release times and higher compression thresholds.

In the present experiment, ratings of acceptance for the *Normal speech & party*-noise fell below the “Tolerable”-category at the shorter RT’s of 40 and 400 ms, when the ratio was 3:1 or greater. On the contrary, acceptance-ratings made at RT = 4000 ms was high in all cases, down to a ratio of 1.5:1. Thus, for a loud speech and noise signal, even at a favourable SNR, the preferred setting seem to be a long release time in combination with a low compression ratio - providing the listener with a realistic loudness for that signal. In case a faster regulation is needed (e.g., when a sudden increase in the input level occurs), the compression ratio should not exceed 3:1, because this will negatively affect the perceived *noisiness* and *clearness of speech* and thereby decrease user-satisfaction (as depicted in fig. 5.12, e, f, g) «««.

6. General discussion

6.1 A fitting objective for the amplification of loud sounds

Several fitting rules for non-linear gain prescription have been based, at least in part, on the principle of loudness normalisation (Killion & Fikret-Pasa, 1993; Cornelisse et al, 1995; Kiessling et al, 1996; Valente et al, 1997; NAL-NL1 by Byrne et al, 2001). The goal for these fitting rules is to amplify all sounds, such that they are perceived by the user as having the same loudness as experienced by the average normal-hearing listener. In regard to the hearing aid processing of loud sounds, it is relevant to ask whether the objective of normalising loudness is the optimal one.

Hearing-aid users listen to speech and sounds that have been sent through an electro-acoustic device. This is inserted at the entrance of the hearing organ, right before the first filter in the filter model described in subsection 1.5 (fig. 1.6). The sound processing taking place in the hearing aid acts as an additional filter in the model, which transforms the sound in order to compensate for the effects of the hearing loss.

For hearing-impaired listeners listening to speech and sounds from the hearing aid, the relevance of listening comfort and sound quality become more prominent. Depending on the sound processing applied at a given input level, the hearing aid user may perceive the sound as being too loud, too soft, too sharp, too dull, etc., as compared to an internal criterion within the listener. This internal criterion is influenced by the perceptual effects caused by the hearing loss, but it is also shaped by individual experience and preference for sound (shown by the second filter in fig. 1.6).

For instance, the hearing aid user may have formerly worked at an industrial plant and be used to loud sounds in the surroundings. Another person may be experienced with listening to music and may be very critical regarding the sound quality of the aid. Also, the psychological state of the listener may have an influence, e.g., if the person is a quiet person who likes his or her surroundings to be quiet. Finally, the listener's interest in the sound also plays an important role, e.g., when listening to loud music or interacting in a conversation at a loud party.

It is this internal criterion that the hearing aid fitting rationale should attempt to meet. This may be a difficult task, as the criterion may differ from person to person. Based on research evidence, the fitting rationale should provide an initial fitting that targets the needs and preferences observed in the large population. Beyond that, the fitting rationale should include subsystems which can be fine-tuned by the dispenser to match the criteria of the individual client. Such subsystems should be focused both on the variability in psychoacoustic measures (e.g., individual variability in loudness perception), and on the individual's preference for sound.

Regarding the amplification of loud sounds, there may not be such a thing as an exact level of *most comfortable loudness* or *upper comfortable loudness*. Rather, a range of levels may be perceived as being comfortable or approaching uncomfortable loudness, as shown by the variability in the data by Pascoe (1988, fig. 2.14). Likewise, the total range of preferred listening levels for soft, medium and loud sounds may depend on the signal, the listening situation, individual user-preference as well as the degree of hearing loss, etc. Therefore, an alternative to the concept of loudness normalisation could be to focus on the variation in level, relative to the *most comfortable loudness* level, that listeners prefer or can accept.

6.2 Suggestions for an amplification strategy for loud sounds, based on results from experiments #1 and #2

In this project, two listening experiments were carried out to investigate the acceptable level variation for loud sounds and the effect of compressor release time for such sounds. In the first experiment, an alternative approach was used to investigate the perception of level fluctuations in loud signals. Four input signals with built-in level-variations of 20 dB were compressed with seven different ratios (1:1 – 10:1) in a simulated slow-acting hearing aid.

It was found that spectral differences among the four signals did influence the hearing-impaired listeners' perceptions of loudness and acceptance when the signals were processed with the same compression ratio. Especially the two signals (1) *Dantale & party*-noise and (2) *Dantale & car*-noise, which differed in noise type and spectral characteristics of the noise, were rated significantly different.

Dantale in broadband party noise received the highest ratings of loudness and the lowest ratings of acceptance, compared to Dantale in low frequency car noise. With the criterion described in subsection 4.4.3, the compression ratio needed to achieve an acceptance rating of "Tolerable" was 2:1 for signal (1) and 1.25 for signal (2). Thus, as the speech signal was the same in both signals, it seems that the noise component determines listeners' gain preferences for the signal in this case.

Such a relationship may have implications for the gain-prescription in non-linear hearing aids. One may imagine a fitting rationale (or algorithm included in a rationale) that regulates the gain for loud sounds, in a way which depends on the input level and spectral characteristics of the signal. When two signals have almost equal RMS-levels and long-term spectra, but one signal contains noise with a broader spectrum, this signal may be given less gain (i.e., by applying a higher compression ratio) compared to signals that contain noise mainly at lower frequencies. This would then keep the user satisfaction at an equal level in both cases.

Such an algorithm could be based on empirical data on the relationship between listeners' preferences for loud sounds and gain settings in hearing aids. An algorithm like this would also require the hearing aid to detect and categorize different noise types and adjust the gain of the compressor in real time.

It should be noted though, that there may not exist a simple relationship between the long-term spectra of noise (as reported in this report) and user acceptance for different signals. Several characteristics of the noise may have an influence, like the peak level, the degree of modulation in the noise and its frequency content, and the issue of whether the noise should be considered an annoying signal or a signal of interest for the user. This issue needs further investigation.

In fact, most commercial hearing aids do regulate the gain depending on the noise levels in separate compression channels. This regulation is the result of noise-reduction algorithms that detect noise on the basis of its non-modulating character. Gain is then lowered in the channels with the greatest noise levels, to avoid masking effects from these channels that might influence overall speech intelligibility. The output response of such a hearing aid with a mixed speech and noise signal is shown in figure 6.1 The effect of such noise reduction systems on speech intelligibility has been questioned, but a positive effect on subjective impressions of listening comfort and sound quality has been found (Boymans & Dreschler, 2000; Alcantara et al, 2003; Ricketts & Hornsby, 2005).

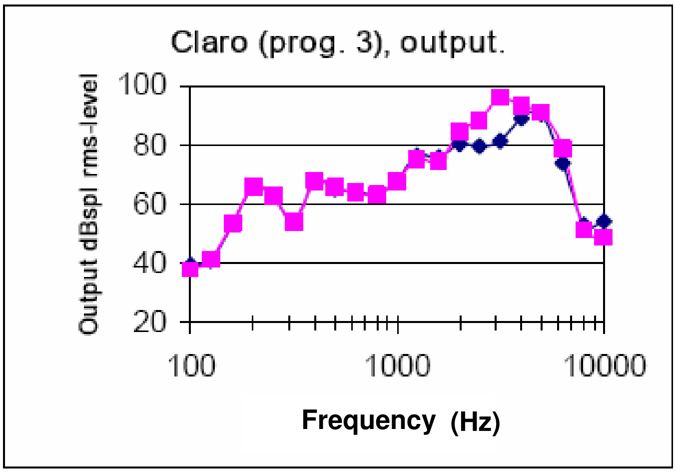

Figure 6.1. Change in the long term output response, from a commercial hearing aid with a channel-specific noise reduction system. In the first condition (■), running speech alone (Dantale) was sent to the hearing aid at an RMS-input level of 65 dB SPL. In the second condition (◆), this signal was mixed with a narrowband noise centred at 3 kHz. The SNR in the region of the noise (1600 – 6300 Hz) was -15.5 dB. It is seen how the hearing aid output is affected specifically in the region of the noise. Output-responses were recorded in a test box (Madsen HAT 500) using an IEC 711 coupler (Schmidt, 2002).

In experiment #1, the compression ratios resulting in an acceptance rating of "Tolerable", yielded a greater loudness than was preferred by subjects for similar input levels in earlier studies (Neuman, 1995b; Smeds, 2004a, 2004b). Even though the studies cannot be directly compared, it is hypothesised that differences in the duration of test signals in the three studies may play a role for the listeners' tolerances of the loud signals. This indicates the need of a subsystem that regulates the gain depending of the duration of the signal, i.e., less gain might be needed for loud sounds of longer duration compared to short loud sounds which may be more acceptable to listeners. One might imagine a system with two compressors, one that uses moderate compression ratios (1.25 – 2:1) for signals with level fluctuations and high sound levels of shorter duration (e.g., 5-10 seconds), and another compressor that applies a higher compression ratio (e.g., 2:1- 5:1) when the input contains high level sounds of longer duration (e.g., 30-45 seconds).

But as shown in experiment #2, the attack and release times used will influence the outcome of such a system. If a high compression ratio (e.g., greater than 3:1) is to be used, the release time should be kept long (e.g., 4000 ms). If a short release time is used (40 ms) in combination with a high compression ratio, the user may experience a reduction in speech clarity and an increased noisiness of the signal that negatively influences user acceptance (as shown in fig. 5.12, subsection 5.4). When a lower compression ratio is used, the influence of the release time diminishes, but still a long release time is shown to produce the highest ratings of speech clarity and acceptance, and the lowest ratings of noisiness. If a shorter release time is needed, the compression ratio should preferably not exceed 3:1.

One drawback of using a long release time in a system that applies less gain for loud sounds of longer duration, is that it takes a relatively long time for that compressor to increase gain in case of a sudden weak sound (e.g., someone speaking at a distance). This is illustrated in situation 1, on the input/gain-function in fig. 3.8 (subsection 3.2), where the gain is stabilized at a reduced level even though the input is a fluctuating signal. This underlines the need for a system that can detect sounds in the user's environment - and only apply the slow regulating system in restricted cases of loud sounds with longer durations (e.g., when using a loud machine or when being in an airplane, where no conversation takes place).

A dual compression system that compensates for variations in the overall level of speech from one situation to another by slowly changing its gain, and with fast regulation that protects users from sudden transients, has been described by Moore et al. (1991). In a later study (Moore et al, 1999), this system was implemented with a compression threshold at 55 dB SPL and a compression ratio in the slow system of 3:1. This system was intended to provide listeners with some impression of the levels of sounds in the environment. Moore and colleagues found that this system was preferred by subjects over systems that primarily used fast time constants. The dual compressor described by Moore et al. cannot be compared to the system proposed here, but it speaks in favour of a system that regulates gain depending on the characteristics of the input signal, whether they are transients or loud sounds of long or short duration.

The dependency of signal duration on the preferred gain for loud sounds should be investigated further in a field study using data logging that is implemented in most digital hearing aids today. A volume control that regulates the gain for high input levels should be implemented in the device. By comparing volume control adjustments with data on signal duration and the spectrum for the given situation, knowledge might be gained concerning whether such a relationship between gain and signal duration exists. Listeners could also be allowed to switch between listening programs with different types of gain regulation for loud input sounds with varying durations. Then, the amount of volume adjustments made by listeners would indicate which program yielded the most preferable setting.

The overall objective for a fitting rationale that regulates gain, depending on the spectral and temporal characteristics of the input-signal, would not be loudness normalisation as such. Instead, the focus would be on listening comfort and on providing the listener with a realistic experience of the level variations in the input signal. As discussed in chapter 1, auditory information about sound levels in the surroundings may be of importance to the listener – both when he or she pays full attention to the sound, but also when only part of the attention is allocated (for instance when listening to traffic sounds, while bicycling in the city).

6.3 Considerations on the test methods used in this project

Laboratory investigations of hearing-aid users perception of real life sounds is complicated by a number of issues. In this section, some issues related to the experimental methods used in this project will be considered.

6.3.1 Issues in the testing of real-life acoustic phenomena in the laboratory

In the anechoic chamber, there are no visual inputs that can provide the listener with the information or distractions present in a real environment. In addition, the influence of the listener's attention to the sound in real life is difficult to recreate in the laboratory.

In the experiments made during this project, no visual indications (such as a screen or pictures) were used. Instead all subjects were instructed about the nature of the listening situations that they were about to hear. They were also explained about the purpose of the test and what considerations they should make when rating on the categorical scales. For instance, in the case of the acceptance scale used in both experiments, subjects were asked to consider how acceptable the level variation in the given signal would be if they should be in that situation for 5-8 minutes.

When giving such detailed instructions, the subject is moved from being a naive listener to an informed listener, and there may be risk of biasing the results of the study. On the other hand, it may also seem relevant to appeal to the listeners' imagination and understanding of the problem under investigation. The experience from this study was that additional time is needed for the subject instruction and that questions should be asked to verify that the listener has understood the task. A training session where the experimenter interacts with the subject is indeed necessary. In experiment #2, a significant effect of the trial was found in the data obtained in one signal on two of the four scales (tables 5.3 and 5.5). Thus, the presentation of six randomly chosen test signals was not enough for subjects to establish a steady criterion on these two scales. Due to time limitations, it was not possible to provide a longer training session, but preferably a full session with 16 signals should have been given. In experiment #1, a full session with 28 signals was given and no trial-effect was seen in this experiment.

The procedure of explaining to subjects the purpose of the listening test may be transferred to the fitting procedure taking place in audiology clinics. Modern digital hearing aids have several advanced features that may be dependent on actions taken by the user (e.g. switching to another program). Often hearing aid users are elderly people, who are not used to the possibilities and operation of digital technology. As was done in this experiment, part of the fitting procedure could include instructions and examples of the purpose and use of specific features in the hearing aid - like it was done in this experiment. If the hearing-aid user understands the purpose of a given feature, he or she will benefit more from the device. The user may also be willing to compromise on certain issues related to a feature (for example, accepting a degraded sound quality in a special noise reduction program, if only speech is clearly intelligible).

6.3.2 Issues related to the categorical scaling method

The categorical scales used in experiments #1 and #2 were prepared with inspiration from earlier studies on sound quality in sound reproduction systems and hearing aids (Gabrielsson 1985, 1979, 1990; Neuman et al, 1998). The scales, *variation*, *loudness*, *clearness of speech*, *noisiness* and *acceptance* were chosen to assess the perceptual effect of the compression parameters under investigation - that is, the compression ratio and the release time. An alternative method would be to first extract the adjectives (or dimensions) that subjects use when they describe their sensations of loud sounds from the hearing aid. This can be done via multidimensional scaling analysis, and this method was used by Gabrielsson and Sjögren (1974, 1975, 1977) who found eight adjectives that were significant for subjects description of sound quality. The adjectives were *loudness*, *clarity*, *fullness*, *spaciousness*, *brightness, softness*, *nearness* and *overall impression*, of which some of these were used in this project.

Instead of categorical scaling with specific adjectives, a comparative procedure could also have been used in this project. In such a test, the subject is presented with two signals (i.e., two signals processed with different compression characteristics) and is asked to select the one he or she prefers the best. After multiple comparisons, the presented signals can be ranked according to their mutual relation. Gabrielsson (1979a) notes that in a comparative procedure, the subject is free to combine several different dimensions in their judgement of the signals. This may be a difficult task, and subjects may fluctuate between dimensions from comparison to comparison. With the categorical scaling procedure, the dimensions are fixed and the subject is “forced” to pay attention to them - even though this requires that the adjectives are relevant for describing the sound, and that the subjects understand the meaning of them.

In experiment #2 of this project, subjects were asked to rate the *clearness of speech*, in the processed versions of the two speech and noise signals. It would also have been relevant to obtain an objective measure of speech intelligibility, e.g., the speech reception threshold for sentences in noise using the Dantale II-test (Wagener et al, 2003). On the other hand, it has been shown that subjects are able to maintain a stable individual criterion of speech intelligibility over several trials (Larsby & Arlinger, 1994; McDaniel & Cox, 1992). Therefore subjective rating of speech intelligibility may be considered a valid tool for selecting the most effective compression settings for speech understanding.

6.3.3 Considerations on the simulated non-linear hearing aid

In both experiments, a simulation of a non-linear hearing aid was attempted. The combination of the compressed signals presented from the loudspeaker and the linear amplification in the hearing aids worn by test subjects, were made to simulate a non-linear device with three compression channels. The compressed signals were amplified by the hearing aids according to the NAL-R procedure (Byrne & Dillon, 1986), such that speech at 62 dB SPL RMS-level would be placed at the subject's most comfortable listening level. The frequency response specified by the NAL-rationale attempts to equalize loudness across frequency, such that all parts of the speech signal contribute equally to its overall loudness.

Depending on the compression ratios and release times applied, the loud input signals used in experiments #1 and #2 would be amplified to the upper part of the hearing impaired listener's audible range. That is, the presentation level was calibrated such that the level variations in the processed signals would occur relative to the output response (baseline) applied to by the hearing aids. The changing slope of the input/output-functions above the anchor-point in figures 4.9 and 5.6 can be thought of as the handle in a commercial hearing aid, varying the degree of gain for high input levels.

The splitting of the processing and presentation of test signals in this experiment was done for three reasons: Firstly, care could be taken in the preparation of test signals such that the output signals from the compressor could be validated and adjusted in level before the actual listening test. Secondly, the linear gain applied in the test situation could easily be individualised for each subject, using the fitting software for the hearing aid. Thirdly, in considering the context of the experiment and the use of categorical scales, listening via hearing aids represented a more realistic situation for the test subjects, e.g. compared to listening to signals under headphones.

In the processing taking place in the MATLAB-compressor and in the presentation of test signals, all signal-levels were referenced to speech at 62 dB SPL RMS-level. The critical point in both experiments was that speech with an input level of 62 dB SPL, being processed through the system, should be perceived by listeners as having a comfortable loudness. In experiment #1, the loudness ratings for speech at 62 dB SPL did not change within signals, but stayed in the range from 3 to 5 - midway between the "soft" and "comfortable" categories (figure 4.18a). A similar rating pattern was found in experiment #2. This verifies that the anchor points of the compressor was adjusted appropriately for a 62 dB SPL speech input, and that the goal of amplifying normal speech to the most comfortable level of the listeners was reached to some extent.

Optimally, mean ratings should have been more centred at the comfortable loudness category, indicating that more gain was needed than prescribed by the NAL-R procedure. Some stretching of the loudness scale may have occurred after all, although it has been found that investigating only part of the loudness function (here the upper part) has no adverse effect on the shape of the obtained functions (Launer, 1995). In all cases, it was satisfying that the loudness of normal speech was not perceived as being higher than comfortable in this study.

The non-linear hearing aid simulated in experiments #1 and #2 does not compare entirely to a real commercial hearing aid. When processing the input signals in the compressor model, the same compression ratio and time constants were used in all three channels, in a given condition. A real hearing aid, fitted according to a loudness normalisation-scheme, would typically contain more channels (e.g., 15 or 20). The compression ratio in individual channels would vary according to the hearing threshold across frequency. Less compression would be needed in frequency regions with mild hearing loss, whereas more would be needed in regions with greater loss, in order to fit the input range into the audible range of the listener. Also, the attack and release-times could vary depending on frequency, input level and the type of signal. This would alter the effective compression ratio and thus the output level of the hearing aid.

Also, regarding the lower part of the input/output-functions (below the anchor-point), no lower compression threshold was implemented in the model, as would be the case in a real hearing aid. This was not considered to be a problem though, because noise in the test signals would fill out the gaps between speech segments, and thus keep the gain from increasing excessively at the higher compression-ratios. In a commercial hearing aid, a lower compression threshold (e.g., at 30 dB SPL), in combination with an expansion segment, would reduce the influence of low level noise.

Generally, the choice of compression settings in this project was made to reduce the number of parameters for investigation. The need for varying the degree of compression across frequency is also reduced by the fact that all subjects had moderately sloping hearing losses. That is, the need for channel dependent compression ratios would be greater in case of steeply sloping loss. Hansen (2002) also argued that an equal setting of the compression ratio across channels can be justified in the case of moderate hearing losses.

6.3.4 Considerations on the test subjects chosen for this study

The eight subjects chosen for this study all had moderately sloping hearing losses. But as seen in figure 4.14, there was some variability in the audiometric configurations. Therefore some differences in the size of the auditory range could also be expected. This might affect the ratings made on the categorical scales. Similarly, differences in previous hearing aid experience (e.g., differences in fitting philosophies) could also affect the data. The confidence intervals seen in the ratings of this study (fig. 4.17-4.19 and fig. 5.11) are, however, relatively small and comparable to the ones reported in similar studies (e.g., Neuman et al, 1998).

The group of hearing-impaired people with mild to moderate losses constitutes a large part of the population with sensorineural hearing loss (Wilson et al, 1999). In cases of greater hearing loss, the residual audible range would be smaller and this would presumably influence the listener’s tolerance for loud signals processed by the hearing aid. A smaller tolerance for loud signals would be expected. It might be that another fitting approach is needed for this group, compared to listeners with moderate and mild losses. Barker et al. (2001) compared wide dynamic range compression with linear amplification in a group of severely hearing impaired subjects. They found that 10 out of 16 subjects preferred a WDRC-scheme with a compression ratio of 2:1, but with a quite high compression threshold (up to 74 dB SPL). Compression may be beneficial also for severe losses, but it should be further investigated how different combinations of compression parameters influence subjective impressions of sound quality and speech intelligibility – also for low and high input levels to the hearing aid.

It should finally be noted that the number of test subjects in this study (8 in experiment #1 and 7 in experiment #2) was quite small. Even though there were significant trends in the obtained data, a greater number of subjects (e.g., 30 or 40) are needed if empirical data should be used as basis for the development algorithms for a hearing aid fitting rule. Also, it would have been beneficial with a group of normal-hearing subjects, in order to compare results and difference in preferences in the two populations. Unfortunately it was not possible to include more subjects in this project, due to time limits «««.

7. Summary and concluding remarks

The aim of this PhD-project was to investigate hearing aid sound processing of loud speech and noise signals, and possibly to suggest amplification strategies for this type of input.

In chapter 1, it was described how the range of speech and sound levels varies over a large span in everyday listening situations. In the data by Wagener et al. (2002), the range between the levels corresponding to the 10 % percentile of the level distribution in the softest sound, and the 90 % percentile of the level distribution in the loudest recording was 58 dB. Most of the sounds recorded by hearing aid listeners had RMS-levels from 60-75 dB SPL and above. Apart from natural sounds in the environment, the hearing aid user will also listen to reproduced speech and sounds, e.g., from radios or when watching a movie. In some cases, the presentation level of the reproduced sound is turned up in order to maintain audibility in the existence of ambient noise.

In chapter 2, it was described how damage to the active cochlear mechanism affects the auditory range and loudness perception in hearing-impaired listeners. The loudness growth function measured via magnitude estimation is steeper in listeners with sensorineural hearing loss. At high sensation levels, it becomes overlapping with the average normal function (denoted as *loudness recruitment*). But as described in subsection 2.3, different degrees of loudness recruitment may exist, yielding either a complete, over- or only partial recruitment in some listeners. Categorical scaling, using verbal loudness categories, has been found to yield a reliable description of loudness sensation, although the obtained functions do not compare directly with functions obtained via magnitude estimation. During the 1990's, several scaling techniques for obtaining individual loudness growth data to be used in non-linear hearing fitting were developed. But comparative studies (Jenstad et al, 1997; Elberling, 1999) have shown great variability in loudness functions obtained with different methods and even within methods due to procedural issues. This has promoted the use of normative data for loudness perception in non-linear gain prescription. Still, categorical scaling techniques may be useful for gathering information from larger groups, for example making relative comparisons of the perceptual effect of different hearing aid settings.

Chapter 3 provided background on the principles for linear and non-linear gain prescription in hearing aids. One commonly used principle is the Wide Dynamic Range Compression (WDRC), in which a larger part of the dynamic input range is compressed and presented within the restricted auditory range of the hearing impaired listener. Subsection 3.1.2.1 contained a description of the NAL-NL1 rationale (Byrne et al, 2001), which aims at amplifying speech to normal loudness, or to a lower than normal loudness if it benefits speech intelligibility. Studies that used categorical scaling of attributes related to sound quality for investigating the perceptual effects of compression have shown a preference for compression ratios no greater than 3:1, when used in combination with long release times. With short release times (e.g., 60 ms) significantly lower ratings of speech clarity, pleasantness and higher rating of increased noisiness have been reported (Neuman et al, 1998).

In studies investigating the preferred listening levels for soft, medium and loud sounds, a preference for presenting such sounds close to the listener's most comfortable loudness level has been found (Neuman et al, 1995b; Smeds et al, 2004a). This would require less gain (or a higher compression ratio) than prescribed by most WDRC-rationales. The question is whether it is preferable to present all sound levels in the range of the most comfortable loudness, or if the fitting objective should rather be to convey auditory information about these level changes to the listener, to the greatest possible extent.

In experiment #1 described in chapter 4, this hypothesis was tested using input signals with built-in level variation. Four different signals containing speech and noise, and noise alone, were processed in a simulated non-linear hearing aid with long attack and release times (AT = 100 ms, RT = 5000 ms). The hearing impaired subjects did perceive the four signals differently, in regard to the perceived level variation, loudness and overall acceptance rated on categorical scales. Differences in noise components and spectral characteristics were hypothesised to play a role for the significant differences in ratings between some of the signals, even though these were sent to the compressor at the same overall RMS-levels. Ratings of acceptance increased with increasing compression ratio, which is in contradiction with the preference found in the literature for ratios no greater than 3:1. A criterion that selected the compression ratio in each ratio yielding a minimum acceptance rating of “Tolerable” was suggested.

In experiments #2 (chapter 5), the perceptual effects of the combination of release time and compression ratio was investigated. Two loud signals, one containing loud speech and noise at 0 dB SNR and the other containing normal speech and noise at +15 dB SNR, were processed with six different compression ratios and three release times. Subjective rated the processed signals in regard to loudness, speech clarity, noisiness and overall acceptance. Using the criterion from experiment #1, with the minimum acceptance rating being “Tolerable”, the best setting was found to be a moderate compression ratio (1.5:1 and 2:1) in combination with the long release time of 4000 ms.

In both experiment #1 and #2, the compression ratios yielding an acceptance rating of “Tolerable” was found to produce a greater loudness rating for the signals, compared to previous studies using similar signals and input levels. This may partly be explained by differences in the methodology used for the listening experiments. Nevertheless, it was hypothesised that differences in the spectra and duration of signals, having identical RMS input levels, should be accounted for by the hearing aid fitting rationale. Thus, the statement made by Moore (1996), that spectral differences among signals plays a lesser role for the estimation of gain targets, due to the lack of loudness summation in hearing impaired listeners (subsection 2.4), cannot be supported.

In chapter 6, suggestions were made for a fitting algorithm that modifies the gain depending on the noise component in the input signal. Also, a system was outlined that utilizes a greater compression ratio for loud sounds of long duration, whereas a lower ratio may be used for signals containing level fluctuations and loud sounds of a short duration. In any case, a long release time (4000 ms) should be used in such systems. If short release times are needed, the compression ratio should not exceed 3:1, as this was found in experiment #2 to generally reduce ratings of speech clarity and give acceptance ratings below the “Tolerable” category.

As outlined in subsection 3.5, it would also have been relevant to investigate the perceptual effects of different frequency responses in combination with the various compression settings. In experiments #1 and #2, the frequency response prescribed by the NAL-R rationale (Byrne & Dillon, 1986) was used as a baseline, providing audibility for the processed signals presented in the free field. The effects of changing the slope of this response might have been investigated in a third experiment. Specifically, a fix point on the response could be chosen (e.g., at 1000 Hz), with the amount of gain being varied at frequencies above and below this point. The hearing impaired listeners’ impressions of loudness, clearness of speech, noisiness and overall acceptance could be obtained via categorical scaling for different input signals with varying noise types, signal-to-noise ratios and speech levels. Also, additional attributes describing the sound quality, like *sharpness* and *fullness* could be investigated.

Keidser et al. (2005) used the paired comparison method to investigate the combined effects of input signal, frequency response, compression settings and listening criteria in 21 hearing-impaired listeners. They used the NAL-RP response as a baseline (Byrne et al, 1991), and made seven alterations from this response with gain reduction at low frequencies and seven alterations with gain reductions at high frequencies (figure 7.1). When the input signal contained noise with a spectrum different from the speech spectrum, listeners preferred the response where frequency regions containing intrusive noise components were dampened. On the contrary, when focus was on speech intelligibility, subjects preferred higher gain even in frequency regions with poor signal-to-noise ratios and high intensity levels.

A similar test methodology might be used for investigating listeners' preferences for frequency response, when the input contains soft and loud speech and noise.

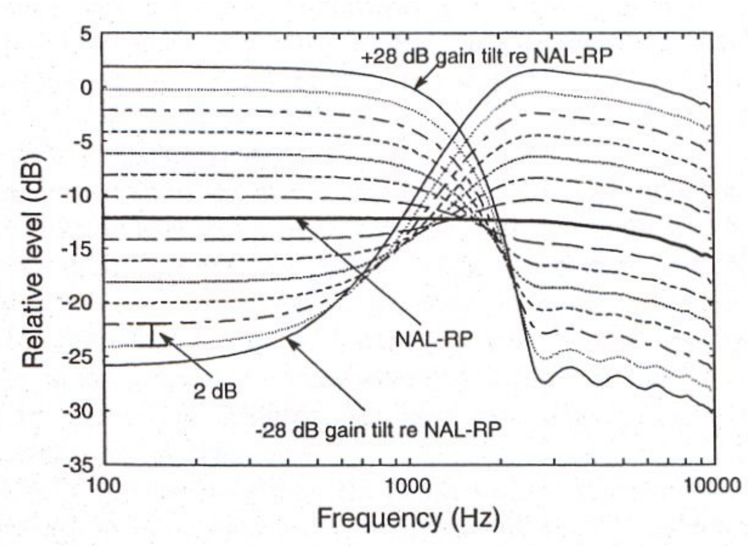

Figure 7.1. Frequency responses used by Keidser et al (2005) in their investigation of the relationship between input signal, hearing aid processing and listening criteria. The solid line indicates the NAL-RP response for the given hearing loss, which acted as a baseline.

In summary, the combination of pre-processed test signals and linear amplification fitted to the individual hearing loss, seems to be a realistic setting for investigating subjective impressions of hearing aid signals presented in the laboratory. But such experiments certainly need to be supplemented by field studies where the user moves around in his or her natural environment.

Apart from research purposes, soft and loud signals might also be used for validation of hearing aid fittings in the clinic. Already, many hearing aid manufactures supply the dispenser with recordings of real-life signals, often as part of the fitting software. Signals containing level variation can be supplemented with other signals, having more steady RMS-levels. While presenting such signals to the client, the clinician can obtain information, either via informal interviewing or by using scales like the ones in the present study. This information may then be used to fine-tune the hearing aid by adjusting the relevant controls regulating the gain in the device. In case the listener feels the loud sounds are annoying, the clinician would lower the value of a specific control, changing underlying parameters in the hearing aid that reduce the overall gain for high input levels «««.

8. References

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9. Appendices

9.1 Audio-CD with input- and test-signals

The input signals to the compressor and the processed test signals presented to subjects in experiment #1 and #2, can be found on the audio-CD attached on page 123. The track list can be found below. Please see chapters 4 and 5 for details.

SNR = signal-to-noise-ratio, **AT** = attack-time, **RT** = release-time, **CR** = compression ratio.

Test signals used in experiment #1

Description	*Track no.*
Input-signals to the compressor	
Signal (1): Dantale speech at normal vocal effort & party-noise (+ 5 dB SNR).	1
Signal (2): Dantale speech at normal vocal effort & car-noise (+ 5 dB SNR).	2
Signal (3): Female and Male speakers at normal, raised and loud vocal effort & party-noise (+ 5 dB SNR).	3
Signal (4): Audience at a football match.	4
Output-signals from the compressor	
Dantale & party-noise, AT100, RT5000, CR1	5
Dantale & party-noise, AT100, RT5000, CR1.25	6
Dantale & party-noise, AT100, RT5000, CR1.5	7
Dantale & party-noise, AT100, RT5000, CR2	8
Dantale & party-noise, AT100, RT5000, CR3	9
Dantale & party-noise, AT100, RT5000, CR5	10
Dantale & party-noise, AT100, RT5000, CR10	11
Dantale & car-noise, AT100, RT5000, CR1	12
Dantale & car-noise, AT100, RT5000, CR1.25	13
Dantale & car-noise, AT100, RT5000, CR1.5	14
Dantale & car-noise, AT100, RT5000, CR2	15
Dantale & car-noise, AT100, RT5000, CR3	16
Dantale & car-noise, AT100, RT5000, CR5	17
Dantale & car-noise, AT100, RT5000, CR10	18
Speakers & party-noise, AT100, RT5000, CR1	19
Speakers & party-noise, AT100, RT5000, CR1.25	20
Speakers & party-noise, AT100, RT5000, CR1.5	21
Speakers & party-noise, AT100, RT5000, CR2	22
Speakers & party-noise, AT100, RT5000, CR3	23
Speakers & party-noise, AT100, RT5000, CR5	24
Speakers & party-noise, AT100, RT5000, CR10	25
Football match, AT100, RT5000, CR1	26
Football match, AT100, RT5000, CR1.25	27
Football match, AT100, RT5000, CR1.5	28
Football match, AT100, RT5000, CR2	29

Football match, AT100, RT5000, CR3	30
Football match, AT100, RT5000, CR5	31
Football match, AT100, RT5000, CR10	32

Test signals used in experiment #2

Description	*Track no.*
Input-signals to the compressor	
Signal (1): Male speaker at loud vocal effort & party noise (0 dB SNR)	33
Signal (2): Male speaker at normal vocal effort & party noise (+15 dB SNR)	34
Output-signals from the compressor	
Loud speech & party-noise, SNR0, AT10, RT40, CR1	35
Loud speech & party-noise, SNR0, AT10, RT40, CR1.5	36
Loud speech & party-noise, SNR0, AT10, RT40, CR2	37
Loud speech & party-noise, SNR0, AT10, RT40, CR3	38
Loud speech & party-noise, SNR0, AT10, RT40, CR5	39
Loud speech & party-noise, SNR0, AT10, RT40, CR10	40
Loud speech & party-noise, SNR0, AT10, RT400, CR1.5	41
Loud speech & party-noise, SNR0, AT10, RT400, CR2	42
Loud speech & party-noise, SNR0, AT10, RT400, CR3	43
Loud speech & party-noise, SNR0, AT10, RT400, CR5	44
Loud speech & party-noise, SNR0, AT10, RT400, CR10	45
Loud speech & party-noise, SNR0, AT10, RT4000, CR1.5	46
Loud speech & party-noise, SNR0, AT10, RT4000, CR2	47
Loud speech & party-noise, SNR0, AT10, RT4000, CR3	48
Loud speech & party-noise, SNR0, AT10, RT4000, CR5	49
Loud speech & party-noise, SNR0, AT10, RT4000, CR10	50
Normal speech & party noise, SNR15, AT10, RT400, CR2	51
Normal speech & party-noise, SNR15, AT10, RT40, CR1.5	52
Normal speech & party-noise, SNR15, AT10, RT40, CR2	53
Normal speech & party-noise, SNR15, AT10, RT40, CR3	54
Normal speech & party-noise, SNR15, AT10, RT40, CR5	55
Normal speech & party-noise, SNR15, AT10, RT40, CR10	56
Normal speech & party-noise, SNR15, AT10, RT400, CR1.5	57
Normal speech & party-noise, SNR15, AT10, RT400, CR3	58
Normal speech & party-noise, SNR15, AT10, RT400, CR5	59
Normal speech & party-noise, SNR15, AT10, RT400, CR10	60
Normal speech & party-noise, SNR15, AT10, RT4000, CR1.5	61
Normal speech & party-noise, SNR15, AT10, RT4000, CR2	62
Normal speech & party-noise, SNR15, AT10, RT4000, CR3	63
Normal speech & party-noise, SNR15, AT10, RT4000, CR5	64
Normal speech & party-noise, SNR15, AT10, RT4000, CR10	65

Audio-CD with input- and test-signals.

See appendix 9.1 for details.

Please contact the author (e.schmidt@widex.com)

to obtain a copy of this CD

9.2 Input and channel levels (anchor-points) used in the compressor-model, in experiments #1 & #2

Experiment #1
The input-gain to the compressor was adjusted such that the RMS input level, measured at the first measurement pin, was 62 dB SPL (or -34 dB re. full scale) for each reference signal. For signal (1), (2) and (3), the reference signal contained speech (with pauses removed) from the 1st segment in each signal. For signal (4), the reference signal contained the audience-noise from the 1st segment in that signal.

The table shows the corresponding RMS-levels in each channel, measured at the second measurement pin in the model. These levels reflect the influence of the attack and release-times on the level detectors in the timing block. These levels were used as anchor-points (that is, the input-levels in each channel that receives the same gain in dB, regardless of the compression ratio).

Channel RMS-levels measured at measurement pin #2, for speech (with pauses removed) at 62 dB SPL RMS-level to the compressor.

Signal	Channel 1	Channel 2	Channel 3
Dantale & party-noise	67.8 dB	58.2 dB	50.1 dB
Dantale & car-noise	67.9 dB	58.2 dB	50.1 dB
Speakers & party-noise	67.5 dB	58.1 dB	52.5 dB
Football match	65.1 dB	63.7 dB	56.5 dB

Experiment #2
The input-gain to the compressor was adjusted such that the RMS input level, measured at the first measurement pin, was 62 dB SPL (or -34 dB re. full scale) for speech at normal vocal effort (with pauses removed).

The table shows the corresponding channel-levels (anchor-points), measured at the second measurement pin in the model.

Channel-levels, measured at the second measurement pin, for the speech (with pauses removed) at 62 dB SPL RMS-level to the compressor.

Signal	Channel 1	Channel 2	Channel 3
Loud speech & party-noise	56.2 dB	41.2 dB	36.4 dB
Normal speech & party-noise	56.2 dB	41.2 dB	36.4 dB

To compensate for the influence of release time, the output RMS-levels for a normal speech input of 62 dB SPL were measured in all combinations of ratios and release-times. Then, in all cases, the level was raised to the output level of the uncompressed condition (1:1). This would simulate a hearing aid that always keeps the same gain for a normal speech input, regardless of changes in compression ratio and time-constants. The dB-values used for compensating the change in output level for each combination of ratio and release-time are shown in the table.

The dB-values used at each compression ratio to raise the output-level from the compressor, in order to compensate for the level-reduction caused by changes in the release time.

	10/40	10/400	10/4000
1.5:1	2.32 dB	0.98 dB	0.31 dB
2:1	3.09 dB	1.65 dB	0.44 dB
3:1	3.60 dB	1.96 dB	0.58 dB
5:1	3.76 dB	2.17 dB	0.59 dB
10:1	3.77 dB	2.26 dB	0.61 dB

9.3 Data-sheets for loudspeakers used in experiments #1 & #2

maj/juni 88

Serie nr.
RRT-A
032632 A
032632 B
RRT-B

Rogers

BRITISH HIGH·FIDELITY

LS3/5A MONITORING LOUDSPEAKER

TECHNICAL SPECIFICATION

SENSITIVITY:	82dB SPL @ 1 metre for 2.83 volts input.
POWER HANDLING:	25 Watts Programme. (This figure is related to the loudspeakers impedance). Suitable for amplifiers from 25–75 watts per channel into 8 ohms.
IMPEDANCE:	15 ohms nominal.
CROSSOVER:	13 Precision elements crossing over 3.0 KHz.
FREQUENCY RESPONSE:	± 3dB 70 Hz – 20 KHz.
DRIVE UNITS:	Selected KEF B110 (110 mm) Bextrene coned Bass/Mid range drive unit. Selected KEF 127 (19 mm) synthetic dome tweeter modified with a special.
DIMENSIONS:	300 mm High. 185 mm Wide. 160 mm Deep.
FINISH:	Walnut (Teak or Black to order). Black Grille Cloth.

HINTS ON USE:

MOUNTING:

For the best results the speaker should be sited on rigid stands approximately 50 cm high, positioned so that side walls or reflecting objects are at least 1 metre away with at least 30 cm space behind the speakers.

The best stereo performance will be achieved with the loudspeakers high frequency drive unit level with the listeners ears, angled inwards so that their axes cross just in front of the listening position.

It is recommended most strongly that before the final position of the loudspeakers is decided upon some experimentation with mounting and positioning is done to ensure the best balance is achieved.

CONNECTORS:

4 mm Banana Posts spaced at 19 mm.

CONNECTION:

As connecting cables can have appreciable insertion losses it is recommended that the following minimum standard of cable is used.

1) Runs of up to 5 m: minimum standard 24/0.2 mm each conductor.
2) Runs of up to 10m: minimum standard 32/0.2 mm each conductor.
3) Runs of over 10 m: minimum standard 50/0.25 mm each conductor.

There are also specialist loudspeaker cables of various qualities widely available which come pre-cut or to order in length.

KEF CAPRICE II

CAPRICE II is a high-quality bookshelf loudspeaker intended for use in living rooms of average size with associated electronics giving up to 100 watts output per channel into 8 ohms. Efficiency is in the medium range at 86dB per watt measured at 1 metre and the system will give peak sound pressure levels of up to 106dB in typical living rooms.

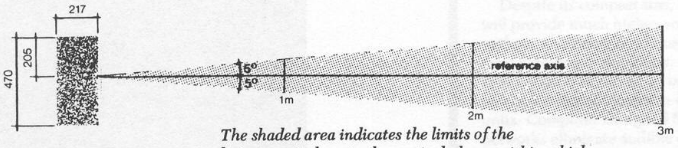
The basic design is a two-way system employing the latest developments in Bextrene-cone drive units to cover the essential musical range with very low colouration. As the originators and pioneers in new diaphragm technology since the early sixties, KEF have acquired unrivalled experience in the design and production of these highly specialised drivers. Bextrene diaphragms, properly damped and terminated, afford the best solution at the present time, being generally more consistent in production and more reliable in use than other fashionably novel materials.

The upper octaves are handled by a 25mm dome radiator with an impeccably smooth frequency response and optimised distribution characteristics. Computer-designed filter sections provide fourth-order target functions centred on 3kHz giving smooth power transference between drive units without the usual disturbing crossover effects.

CAPRICE II is available in simulated walnut veneer with removable textile grille.

SPECIFICATION

Frequency range	68Hz to 20kHz ±2.5dB at 2m on reference axis (−10dB at 42Hz and 30kHz)
Directional characteristics	within 2dB of response on reference axis up to 20kHz for ±5° vertically up to 10kHz for ±20° horizontally
Maximum output	106dB spl on programme peaks under typical listening conditions
Characteristic sensitivity level	86dB spl at 1m on reference axis for pink noise input of 1W
Distortion	Measured at 1m on reference axis at mean spl of 90dB, anechoic conditions Second harmonic: less than 1% from 150Hz to 20kHz Third harmonic: less than 1% from 50Hz to 20kHz
Enclosure type	Closed box with third-order 1f attenuation characteristic
Internal volume	19 litres
Nominal impedance	8 ohms
Programme rating	100W
Maximum continuous sinusoidal input	20V rms from 20Hz to 2.5kHz reducing to 8V rms from 4kHz to 20kHz
Minimum amplifier requirements	15W
Weight	8.3kg (18lb)
Dimensions	470 (h) × 280 (w) × 217mm (d) 18½ (h) × 11 (w) × 8½in (d)



The shaded area indicates the limits of the listening window, in the vertical plane, within which optimum tonal balance and stereophonic effects will be perceived.

KEF reserves the right to incorporate developments and amend the specifications without prior notice, in line with continuous research and development.

KEF

KEF products are manufactured in England and distributed in the United Kingdom by:
KEF Electronics Ltd
Tovil
Maidstone
Kent ME15 6QP England
Telephone: Maidstone (0622) 672261
Telex: 96140

Distribution in the USA by:
Intratec
PO Box 17414
Dulles International Airport
Washington, DC 20041 USA
Telephone: (703) 435 9100

Printed in England

Part No. PL 307 EN 01

9.4 Data-sheet for Widex Senso Diva SD9M BTE hearing aid

SD-9 (SD-9M)

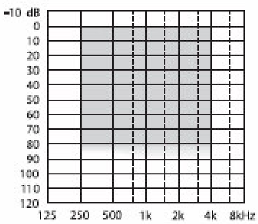
Senso Diva 100% digital BTE

- Sensogram
- Diva Compression
- Diva Noise Reduction with Speech Intensification
- Diva Locator
- Diva Feedback Cancelling
- Diva Occlusion Manager

Recommended for:

- Mild and moderate-to-severe losses
- Flat, sloping, ski-slope and reverse slope losses
- Cookie-bite losses

Suggested Fitting Range



Senso Diva BTEs offer cutting edge technology and the most advanced audiological solutions in a very attractive package.

- 2MHz/1bit sigma-delta converters
- 32 kHz/20 bit processing
- Sensogram in 4 or 13 bands as required
- Enhanced Dynamic Range Compression in 15 channels, Sound Stabilizer and Anti Smearing System
- Diva Noise Reduction with Speech Intensification in 15 channels
- Diva Locator is a new dual microphone system with Adaptive Beamforming, OptiMic system with adaptive matching and Noise Classification
- Diva Feedback Cancelling with Feedback Path Simulator and Dynamic Cancellation Optimiser
- Diva Occlusion Manager with fine tuning of own voice perception
- Beep-tone indicator for programme selection and low battery
- Easy programming with NOAH/Compass or the SP3 dedicated programmer
- Choice between listening programmes: M, MT, T (and Music for SD-9M)
- Optional volume control

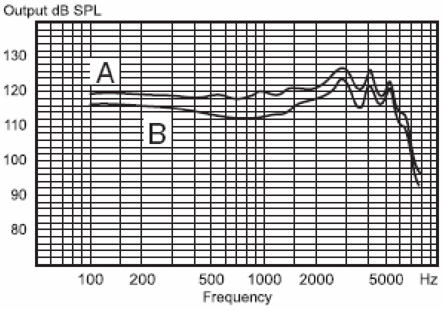
Technical Data

	711 Ear simulator	2cc Coupler
OSPL90: Peak	127 dB SPL	118 dB SPL
1000 Hz	120 dB SPL	113 dB SPL
HAIC	120 dB SPL	113 dB SPL
Battery Drain (st.by)	1.15 mA	
Battery Drain	1.15 mA	
Battery Type 13 Zn-Air (270 mAh)	230 hours	
Telecoil TLS*	+ 2 dB	
Harmonic Distortion	1%	
IRIL (GSM/DCS interference level)	5/15 dB SPL	

* *A telecoil input of 100 mA/m will equal a microphone input of 70 dB SPL.*

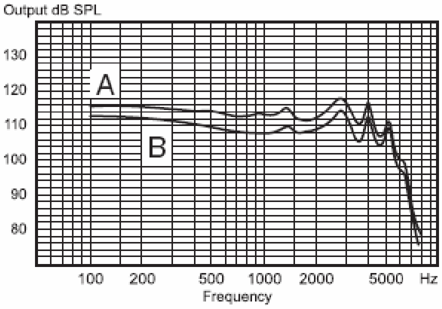
AOC (Automatic Output Control) is an output limiting compression circuit that eliminates distortion from saturation. It can be turned on (=factory setting) and off from the programming devices.

Maximum output (ear simulator - IEC711)



A: AOC disabled B: AOC enabled

Maximum output (2cc coupler - IEC126)



A: AOC disabled B: AOC enabled

Sampling rate	32 kHz
Max word length	32 bits
A/D Converters	2MHz/1bit sigma-delta
DDD stage	1MHz/1bit sigma-delta
System delay	<2 msec
Processor type	Dedicated ASIC
Frequency bands	15 in 1/3 octaves
Channels	15

9.5 Hearing aid fitting-data and test-box measurements of gain and output characteristics

9.5.1 HA-data for subject BN

Insertion gain-values adjusted to NAL-RP targets, are shown in the “General fine tuning data” (the parameters: *IGnormal*, *IGloud* and *IGsoft*)

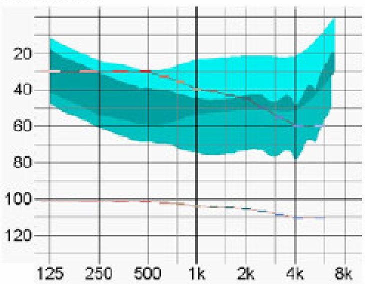
WIDEX
high definition hearing

COMPASS

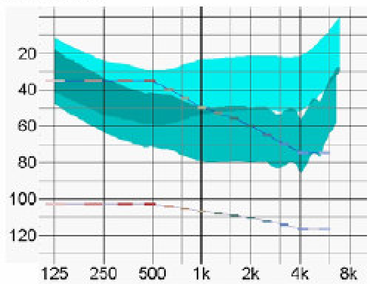
Birth date:		Client no.:	0000003
Last name:		Print date:	12-07-2006
First name:		Printed by:	ABC

Right-ear hearing aid:	Diva SD-9M	Left-ear hearing aid:	Diva SD-9M
Serial number:	170220	Serial number:	170204
Fitting rationale:	Adult/Child > 5 years	Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Speech Spectra (normal)



Right ear

Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	30	40	45	60
Air-bone gap	---	---	---	---
Sensogram HTL	30	40	45	60
Fb Test	---	---	---	---
Available Gain	---	---	---	---
Fb Cancelling		On		
Fb Margin		6		

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	8	14	12	13
IGnormal	! 7	! 19	! 19	! 22
Target	1	7	6	6
IGloud	! 7	! 19	! 19	! 21
Target	15	22	23	30
IGsoft	! 7	! 20	! 19	! 23

Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Left ear

Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	35	50	60	75
Air-bone gap	---	---	---	---
Sensogram HTL	35	50	60	75
Fb Test	---	---	---	---
Available Gain	---	---	---	---
Fb Cancelling		On		
Fb Margin		6		

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	11	18	17	22
IGnormal	! 10	! 24	! 25	! 29
Target	4	11	10	15
IGloud	! 10	! 24	! 25	! 29
Target	19	27	31	39
IGsoft	! 10	! 24	! 25	! 30

Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Compass V3.4.1 - 1129 - Danmark

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

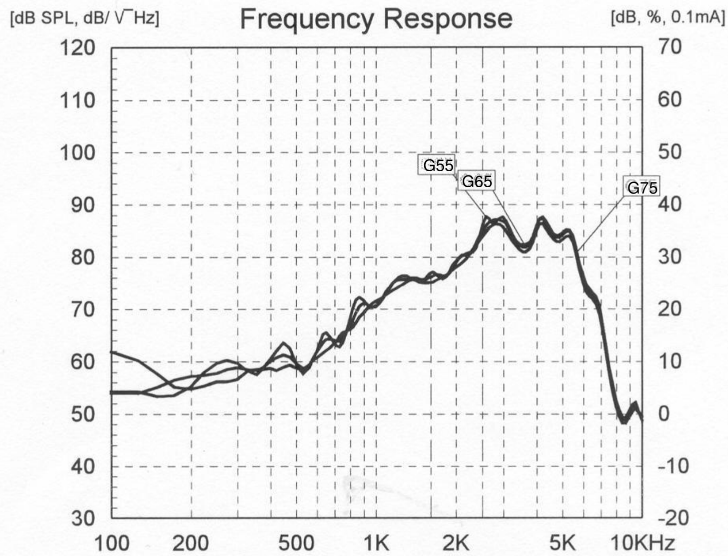
(subject BN, right hearing aid)

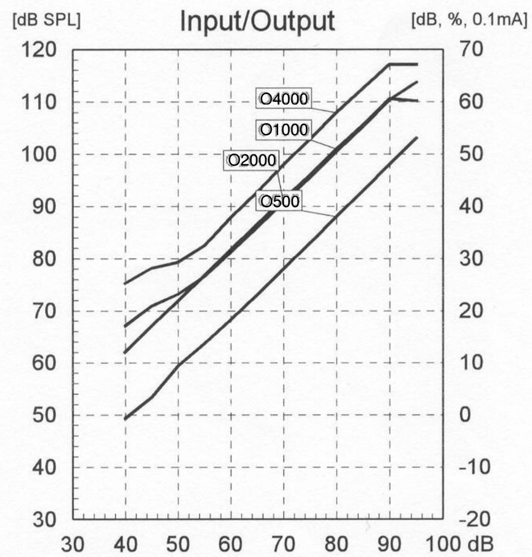
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: BN
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** Right

Peak Gain 1: Input: 55 dB (WN)	38 dB at 2580 Hz
Peak Gain 2: Input: 65 dB (WN)	37 dB at 4250 Hz
Peak Gain 3: Input: 75 dB (WN)	36 dB at 4250 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

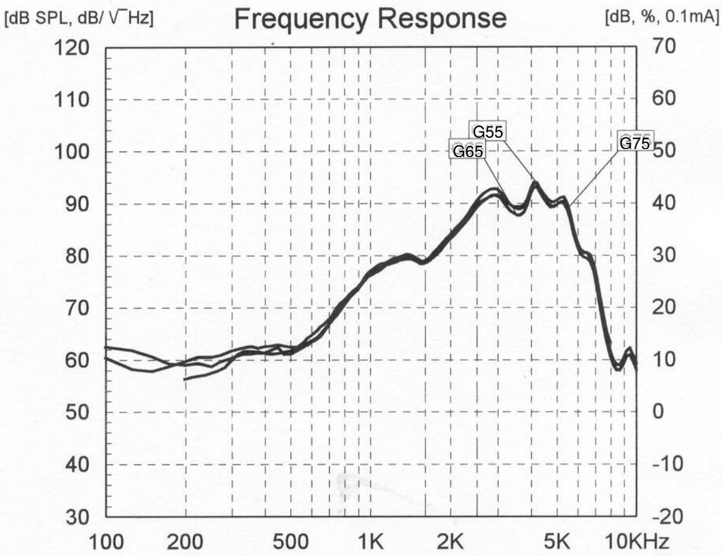
(subject BN, left hearing aid)

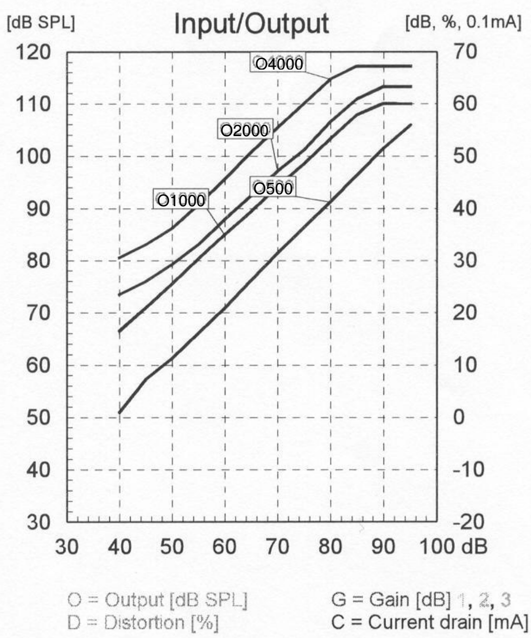
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: BN
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** ~~Right~~ Left

Peak Gain 1: Input: 55 dB (WN)	44 dB at 4250 Hz
Peak Gain 2: Input: 65 dB (WN)	43 dB at 4120 Hz
Peak Gain 3: Input: 75 dB	45 dB at 4120 Hz





9.5.2 HA-data for subject EA

Insertion gain-values adjusted to NAL-RP targets, are shown in the “General fine tuning data” (the parameters: *IGnormal*, *IGloud* and *IGsoft*)

WIDEX
high definition hearing

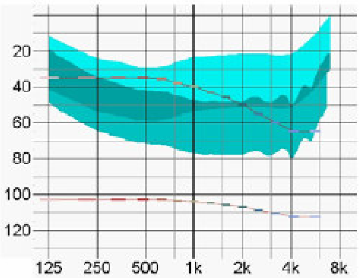
COMPASS

Birth date:	
Last name:	
First name:	

Client no.:	0000002
Print date:	12-07-2006
Printed by:	ABC

Right-ear hearing aid:	Diva SD-9M
Serial number:	170220
Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	35	40	50	65
Air-bone gap	---	---	---	---
Sensogram HTL	35	40	50	65
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling	On
Fb Margin	6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	9	15	14	16
IGnormal	! 10	! 22	! 23	! 23
Target	3	8	7	9
IGloud	! 10	! 22	! 22	! 23
Target	18	22	26	33
IGsoft	! 10	! 22	! 23	! 24

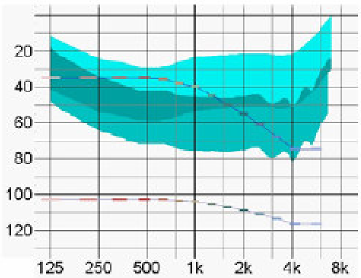
Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Left-ear hearing aid:	Diva SD-9M
Serial number:	170204
Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	35	40	55	75
Air-bone gap	---	---	---	---
Sensogram HTL	35	40	55	75
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling	On
Fb Margin	6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	10	15	15	21
IGnormal	! 11	! 20	! 21	! 25
Target	3	8	8	14
IGloud	! 11	! 20	! 21	! 25
Target	18	23	29	38
IGsoft	! 11	! 21	! 22	! 26

Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Compass V3.4.1 - 1129 - Danmark

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

(Subject EA, right hearing aid)

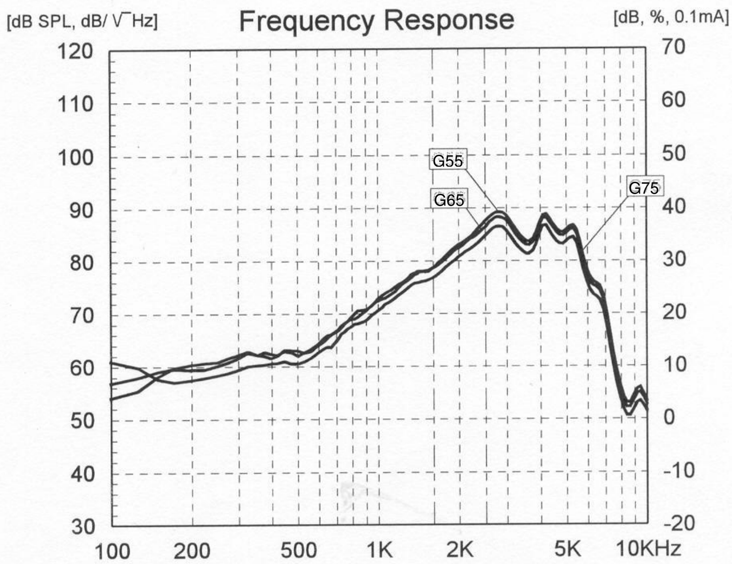
Hearing Instrument Test Results 12-07-06

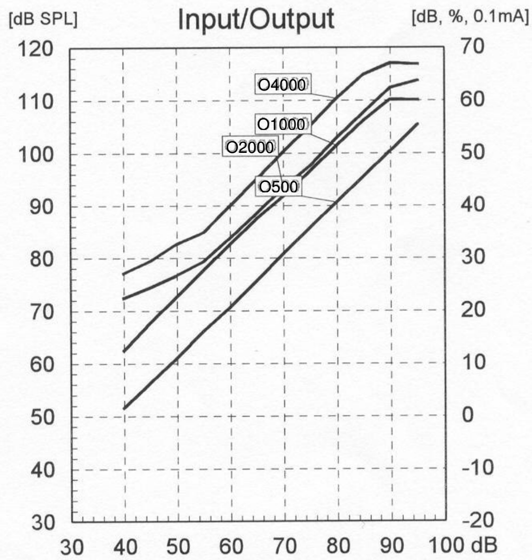
Non-standard test results

Client: ~~BN~~ EA

ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** Right

Peak Gain 1: Input: 55 dB (WN)	39 dB at 2800 Hz
Peak Gain 2: Input: 65 dB (WN)	38 dB at 2800 Hz
Peak Gain 3: Input: 75 dB (WN)	37 dB at 4250 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3

D = Distortion [%] C = Current drain [mA]

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

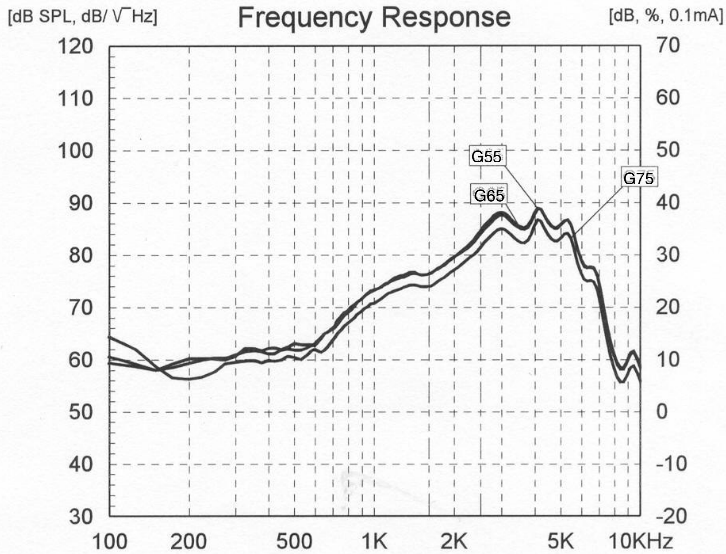
(Subject EA, left hearing aid)

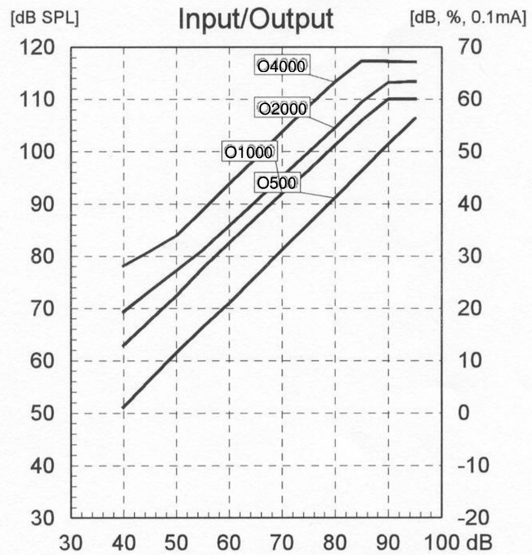
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: ~~BN~~ EA
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** ~~Right~~ Left

Peak Gain 1: Input: 55 dB (WN)	39 dB at 4120 Hz
Peak Gain 2: Input: 65 dB (WN)	39 dB at 4120 Hz
Peak Gain 3: Input: 75 dB (WN)	37 dB at 4120 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

9.5.3 HA-data for subject AJ

Insertion gain-values adjusted to NAL-RP targets, are shown in the "General fine tuning data" (the parameters: *IGnormal*, *IGloud* and *IGsoft*)

WIDEX
high definition hearing

COMPASS

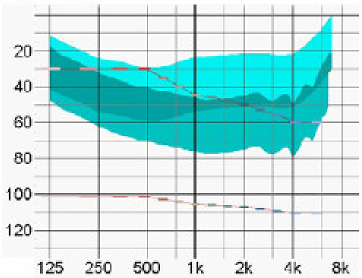
Birth date:	
Last name:	
First name:	

Client no.:	0000006
Print date:	12-07-2006
Printed by:	ABC

Right ear

Right-ear hearing aid:	Diva SD-9M
Serial number:	170220
Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	30	45	50	60
Air-bone gap	---	---	---	---
Sensogram HTL	30	45	50	60
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling	On
Fb Margin	6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	8	16	14	14
IGnormal	8	! 21	! 21	! 23
Target	1	9	7	7
IGloud	! 8	! 21	! 21	! 22
Target	16	25	26	30
IGsoft	! 9	! 22	! 21	! 23

Options

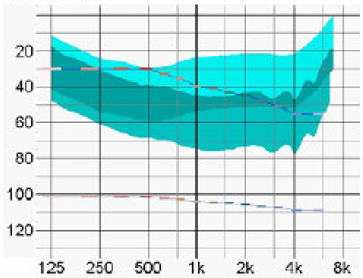
Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Left ear

Left-ear hearing aid:	Diva SD-9M
Serial number:	170204
Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	30	40	45	55
Air-bone gap	---	---	---	---
Sensogram HTL	30	40	45	55
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling	On
Fb Margin	6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	8	14	12	12
IGnormal	! 7	! 19	! 19	! 21
Target	1	7	6	5
IGloud	! 7	! 19	! 19	! 20
Target	15	22	23	28
IGsoft	! 7	! 20	! 19	! 22

Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Compass V3.4.1 - 1129 - Danmark

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

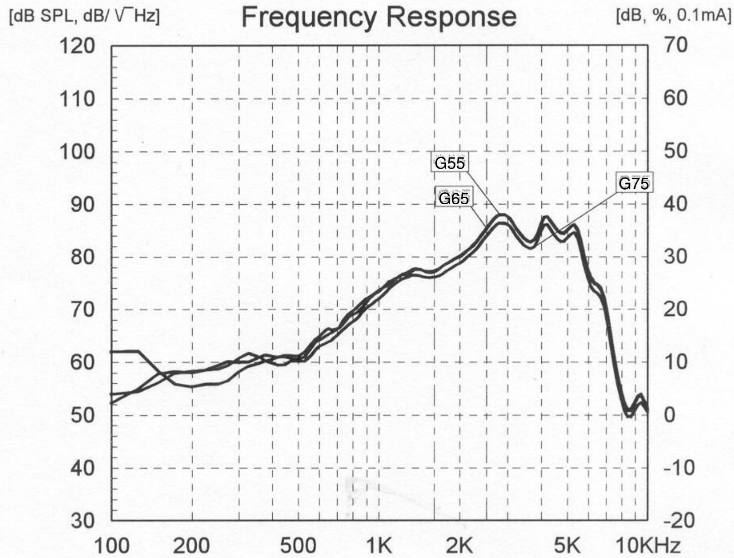
(Subject AJ, right hearing aid)

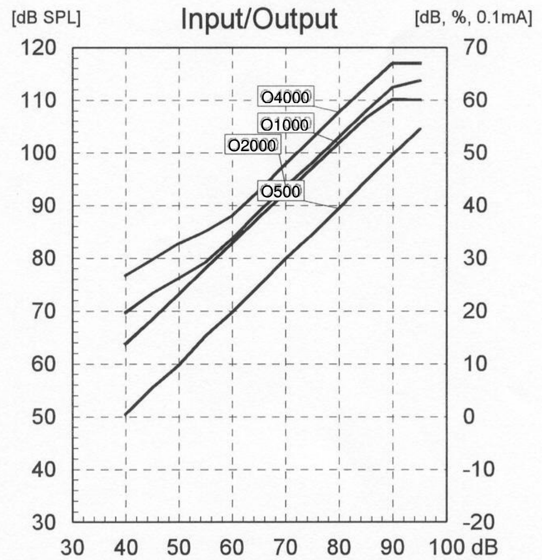
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: AJ.
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** Right

Peak Gain 1: Input: 55 dB (WN)	38 dB at 2800 Hz
Peak Gain 2: Input: 65 dB (WN)	38 dB at 2900 Hz
Peak Gain 3: Input: 75 dB (WN)	36 dB at 2800 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

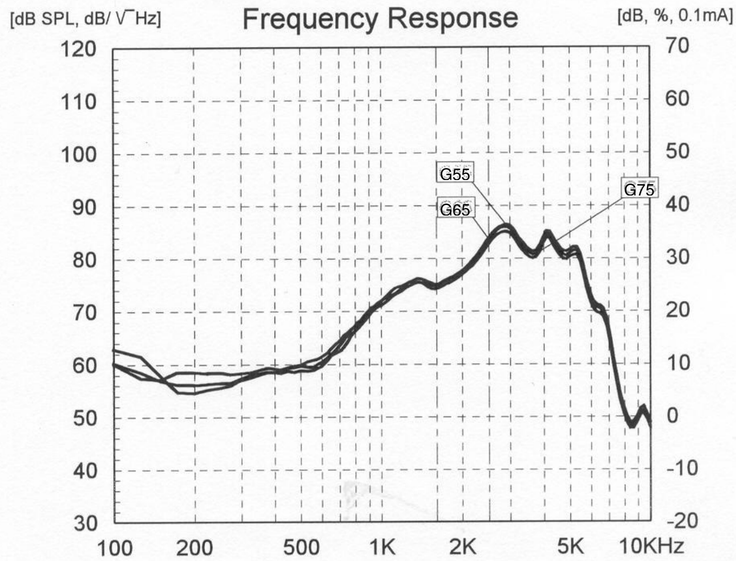
(Subject AJ, left hearing aid)

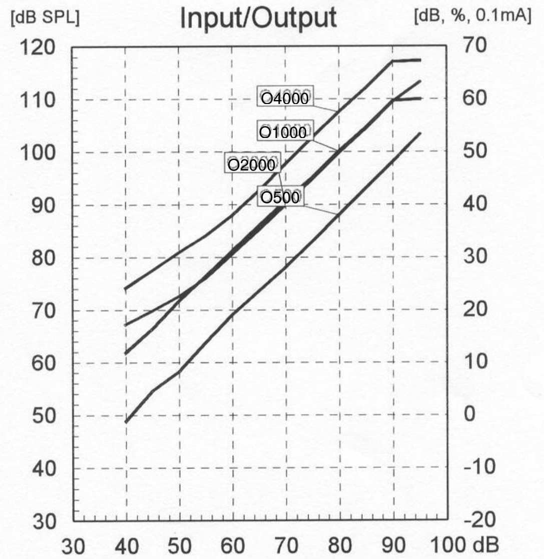
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: ~~BN~~ AJ
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** ~~Right~~ Left

Peak Gain 1: Input: 55 dB (WN)	36 dB at 2900 Hz
Peak Gain 2: Input: 65 dB (WN)	36 dB at 2900 Hz
Peak Gain 3: Input: 75 dB (WN)	35 dB at 2900 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

9.5.4 HA-data for subject BH

Insertion gain-values adjusted to NAL-RP targets, are shown in the "General fine tuning data" (the parameters: *IGnormal*, *IGloud* and *IGsoft*)

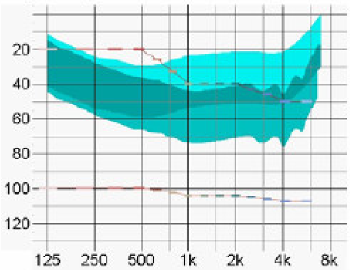
WIDEX
high definition hearing

COMPASS

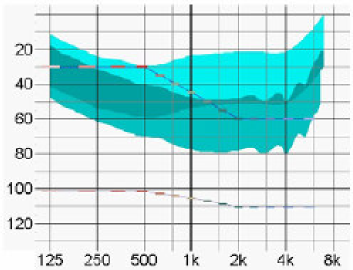
Birth date:		Client no.:	0000012
Last name:		Print date:	12-07-2006
First name:		Printed by:	ABC

Right-ear hearing aid:	Diva SD-9M	Left-ear hearing aid:	Diva SD-9M
Serial number:	170220	Serial number:	170204
Fitting rationale:	Adult/Child > 5 years	Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Speech Spectra (normal)



Fitting data (right ear)

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	20	40	40	50
Air-bone gap	---	---	---	---
Sensogram HTL	20	40	40	50
Fb Test	---	---	---	---
Available Gain	---	---	---	---
Fb Cancelling		On		
Fb Margin		6		

Fitting data (left ear)

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	30	45	60	60
Air-bone gap	---	---	---	---
Sensogram HTL	30	45	60	60
Fb Test	---	---	---	---
Available Gain	---	---	---	---
Fb Cancelling		On		
Fb Margin		6		

General fine tuning data (right ear)

	500Hz	1000Hz	2000Hz	4000Hz
Target	4	13	11	10
IGnormal	! 3	! 18	! 16	! 19
Target	-2	6	4	3
IGloud	! 3	! 18	! 16	! 18
Target	10	21	21	25
IGsoft	! 4	! 18	! 17	! 20

General fine tuning data (left ear)

	500Hz	1000Hz	2000Hz	4000Hz
Target	9	16	16	14
IGnormal	! 8	! 22	! 24	! 23
Target	2	10	9	7
IGloud	! 8	! 22	! 24	! 22
Target	16	25	31	31
IGsoft	! 8	! 23	! 24	! 24

Options

	Right	Left
Occlusion Manager	Off	Off
LF1		
LF2		
PRG	1: M M+T Tele MUS	1: M M+T Tele MUS
Music program gain	0	0
LCT	OMNI	OMNI
AOC	On	On
Telegain	0	0

AIS

AIS

Compass V3.4.1 - 1129 - Danmark

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

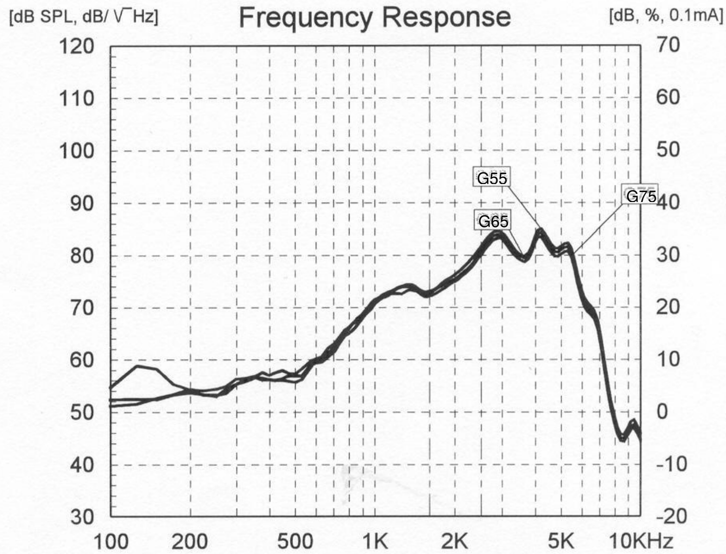
(Subject BH, right hearing aid)

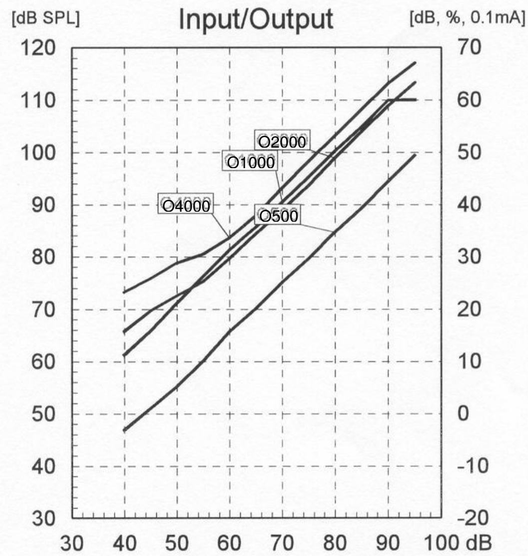
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: ~~BN~~ BWH
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** Right

Peak Gain 1: Input: 55 dB (WN)	35 dB at 4250 Hz
Peak Gain 2: Input: 65 dB (WN)	34 dB at 4250 Hz
Peak Gain 3: Input: 75 dB (WN)	33 dB at 4120 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

(Subject BH, left hearing aid)

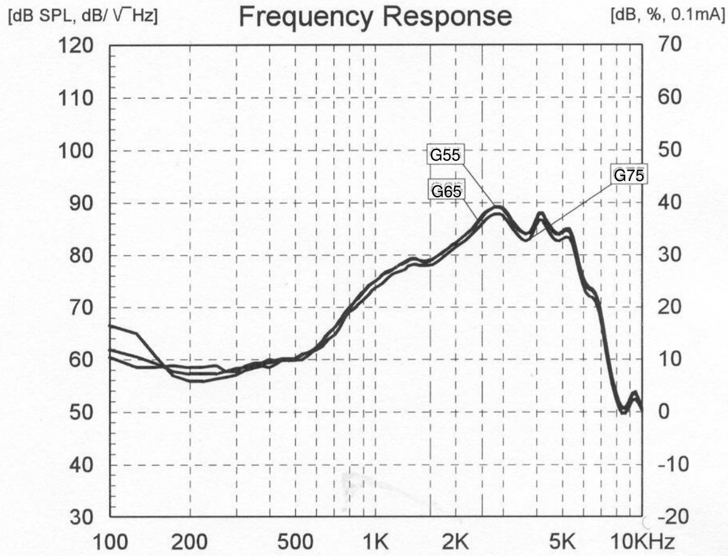
Hearing Instrument Test Results 12-07-06

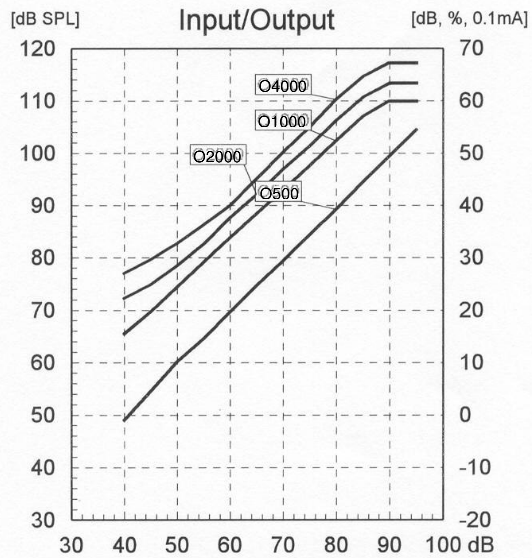
Non-standard test results

Client: ~~BN~~ BWH

ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** ~~Right~~ Left

Peak Gain 1: Input: 55 dB (WN)	39 dB at 2800 Hz
Peak Gain 2: Input: 65 dB (WN)	39 dB at 2800 Hz
Peak Gain 3: Input: 75 dB (WN)	38 dB at 2800 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3

D = Distortion [%] C = Current drain [mA]

9.5.5 HA-data for subject JG

Insertion gain-values adjusted to NAL-RP targets, are shown in the “General fine tuning data” (the parameters: *IGnormal*, *IGloud* and *IGsoft*)

WIDEX high definition hearing

COMPASS

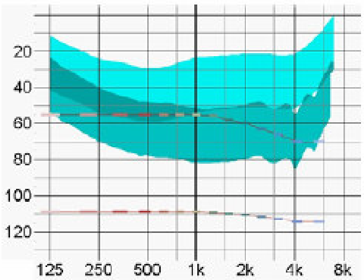
Birth date: [redacted]
Last name:
First name:

Client no.: 0000013
Print date: 12-07-2006
Printed by: ABC

Right ear

Right-ear hearing aid: Diva SD-9M
Serial number: 170220
Fitting rationale: Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	55	55	60	70
Air-bone gap	---	---	---	---
Sensogram HTL	55	55	60	70
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling On
Fb Margin 6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	17	20	18	21
IGnormal	! 16	! 26	! 24	! 29
Target	10	13	11	14
IGloud	! 16	! 26	! 24	! 29
Target	30	31	32	38
IGsoft	! 17	! 27	! 25	! 29

Options

Occlusion Manager Off
LF1
LF2

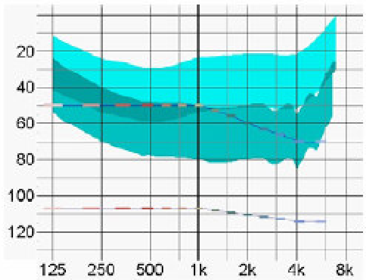
PRG 1: M M+T Tele MUS
Music program gain 0
LCT OMNI
AOC On
Telegain 0

AIS

Left ear

Left-ear hearing aid: Diva SD-9M
Serial number: 170204
Fitting rationale: Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	50	50	60	70
Air-bone gap	---	---	---	---
Sensogram HTL	50	50	60	70
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling On
Fb Margin 6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	15	19	18	20
IGnormal	! 16	! 25	! 26	! 28
Target	8	11	11	13
IGloud	! 16	! 24	! 26	! 28
Target	27	28	32	37
IGsoft	! 17	! 25	! 26	! 29

Options

Occlusion Manager Off
LF1
LF2

PRG 1: M M+T Tele MUS
Music program gain 0
LCT OMNI
AOC On
Telegain 0

AIS

Compass V3.4.1 - 1129 - Danmark

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

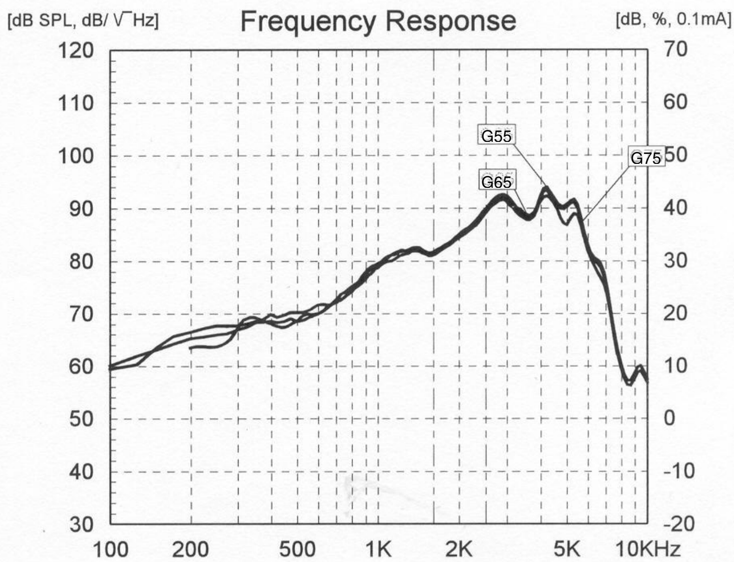
(Subject JG, right hearing aid)

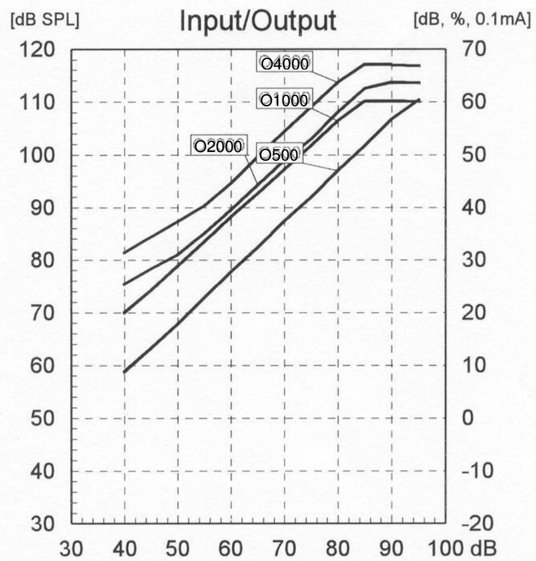
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: ~~BN~~ JG
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** Right

Peak Gain 1: Input: 55 dB (WN)	44 dB at 4250 Hz
Peak Gain 2: Input: 65 dB (WN)	43 dB at 4250 Hz
Peak Gain 3: Input: 75 dB	43 dB at 4250 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

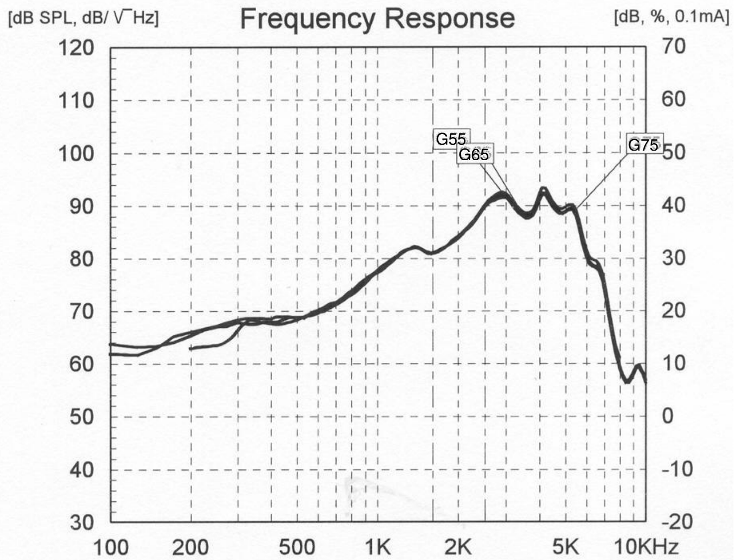
(Subject JG, left hearing aid)

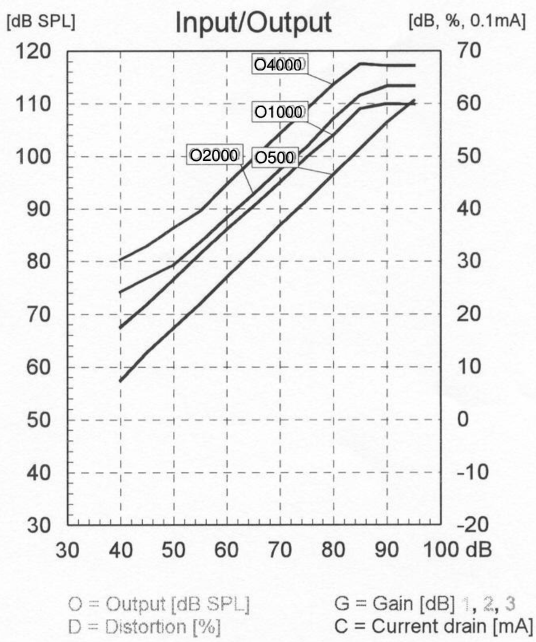
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: JG
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** ~~Right~~ Left

Peak Gain 1: Input: 55 dB (WN)	43 dB at 2900 Hz
Peak Gain 2: Input: 65 dB (WN)	42 dB at 4120 Hz
Peak Gain 3: Input: 75 dB	44 dB at 4120 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

9.5.6 HA-data for subject MA

Insertion gain-values adjusted to NAL-RP targets, are shown in the “General fine tuning data” (the parameters: *IGnormal*, *IGloud* and *IGsoft*)

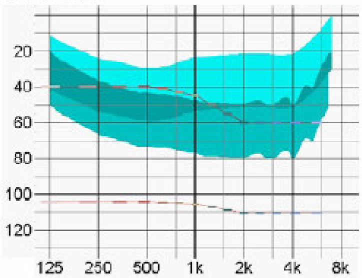
WIDEX high definition hearing

COMPASS

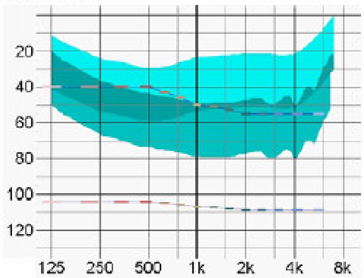
Birth date:	[redacted]	Client no.:	0000008
Last name:		Print date:	12-07-2006
First name:		Printed by:	ABC

Right-ear hearing aid:	Diva SD-9M	Left-ear hearing aid:	Diva SD-9M
Serial number:	170220	Serial number:	170204
Fitting rationale:	Adult/Child > 5 years	Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Speech Spectra (normal)



Right ear

Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	40	45	60	60
Air-bone gap	---	---	---	---
Sensogram HTL	40	45	60	60
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling: On
Fb Margin: 6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	12	17	17	15
IGnormal	12	! 22	! 25	! 24
Target	5	10	10	8
IGloud	! 12	! 22	! 25	! 23
Target	21	25	31	31
IGsoft	! 12	! 22	! 25	! 24

Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Left ear

Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	40	50	55	55
Air-bone gap	---	---	---	---
Sensogram HTL	40	50	55	55
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling: On
Fb Margin: 6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	12	18	16	14
IGnormal	12	! 24	! 23	! 24
Target	5	11	9	7
IGloud	! 12	! 24	! 23	! 22
Target	21	27	29	29
IGsoft	! 12	! 24	! 23	! 25

Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Compass V3.4.1 - 1129 - Danmark

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

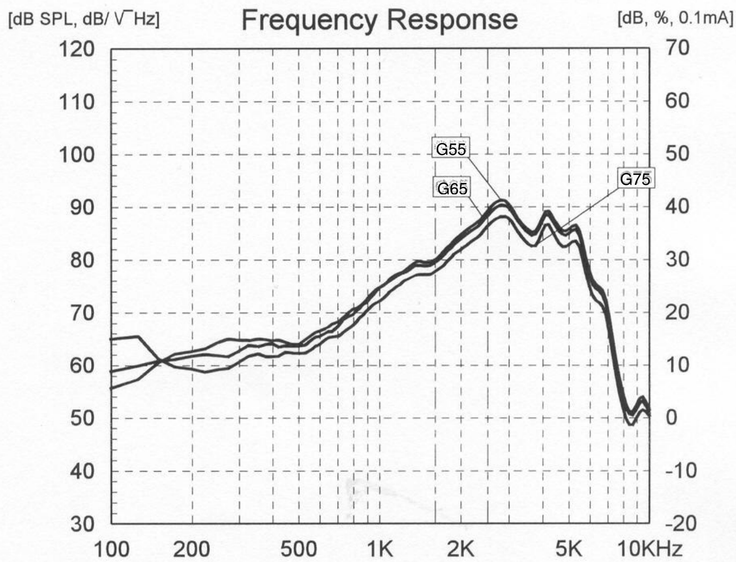
(Subject MA, right hearing aid)

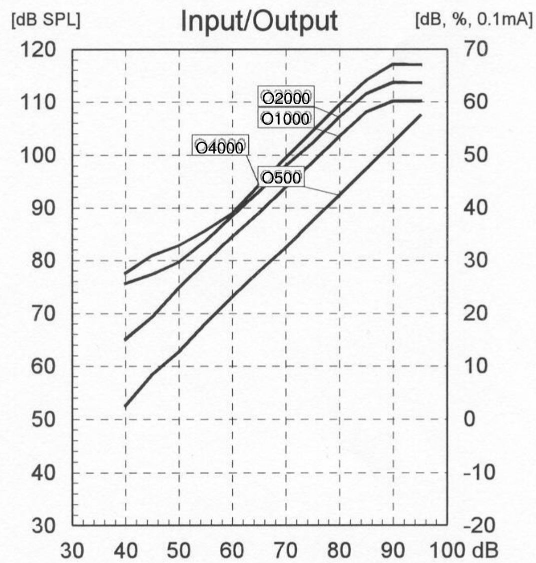
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: MA
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** Right

Peak Gain 1: Input: 55 dB (WN)	41 dB at 2800 Hz
Peak Gain 2: Input: 65 dB (WN)	40 dB at 2800 Hz
Peak Gain 3: Input: 75 dB (WN)	38 dB at 2800 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

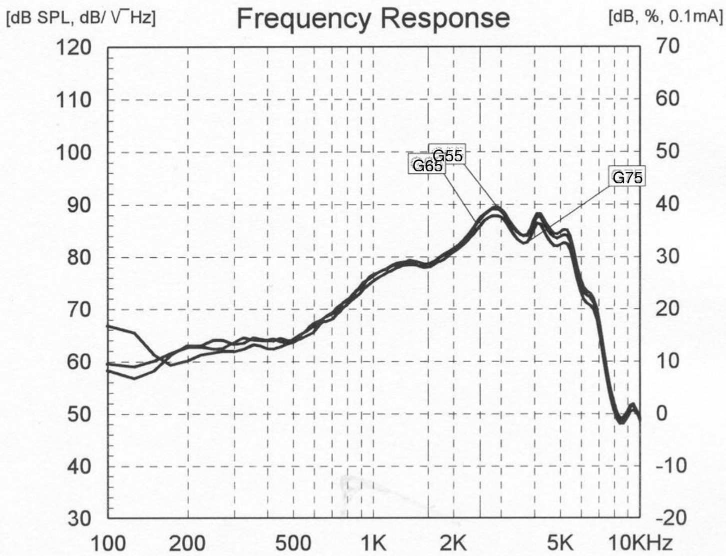
(Subject MA, left hearing aid)

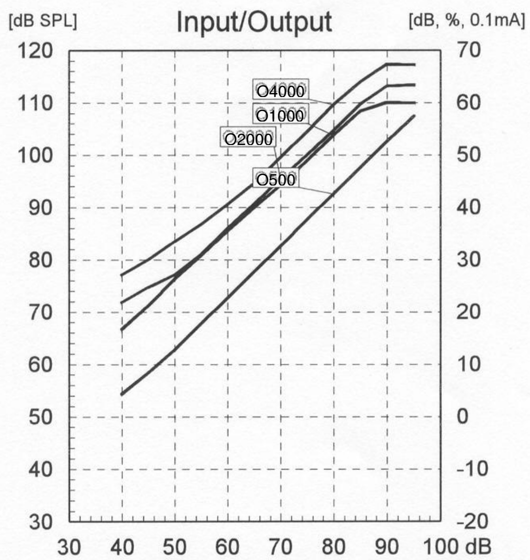
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: ~~BN~~ MA
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** ~~Right~~ Left

Peak Gain 1: Input: 55 dB (WN)	40 dB at 2900 Hz
Peak Gain 2: Input: 65 dB (WN)	39 dB at 2800 Hz
Peak Gain 3: Input: 75 dB (WN)	38 dB at 2800 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

9.5.7 HA-data for subject LKH

Insertion gain-values adjusted to NAL-RP targets, are shown in the "General fine tuning data" (the parameters: *IGnormal*, *IGloud* and *IGsoft*)

WIDEX high definition hearing

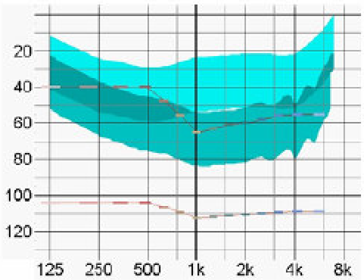
COMPASS

Birth date:
Last name:
First name:

Client no.: 0000018
Print date: 12-07-2006
Printed by: ABC

Right-ear hearing aid: Diva SD-9M
Serial number: 170220
Fitting rationale: Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	40	65	60	55
Air-bone gap	---	---	---	---
Sensogram HTL	40	65	60	55
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling: On
Fb Margin: 6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	13	23	18	16
IGnormal	13	! 29	! 26	! 23
Target	6	16	11	9
IGloud	! 13	! 29	! 26	! 23
Target	22	36	32	29
IGsoft	! 14	! 30	! 27	! 23

Options

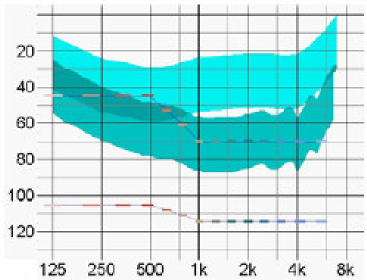
Occlusion Manager: Off
LF1
LF2

PRG: 1: M M+T Tele MUS
Music program gain: 0
LCT: OMNI
AOC: On
Telegain: 0

AIS

Left-ear hearing aid: Diva SD-9M
Serial number: 170204
Fitting rationale: Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	45	70	70	70
Air-bone gap	---	---	---	---
Sensogram HTL	45	70	70	70
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling: On
Fb Margin: 6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	16	27	24	22
IGnormal	16	! 32	! 30	! 29
Target	9	19	17	15
IGloud	! 16	! 32	! 30	! 29
Target	25	39	39	39
IGsoft	! 16	! 32	! 31	! 30

Options

Occlusion Manager: Off
LF1
LF2

PRG: 1: M M+T Tele MUS
Music program gain: 0
LCT: OMNI
AOC: On
Telegain: 0

AIS

Compass V3.4.1 - 1129 - Danmark

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

(Subject LKH, right hearing aid)

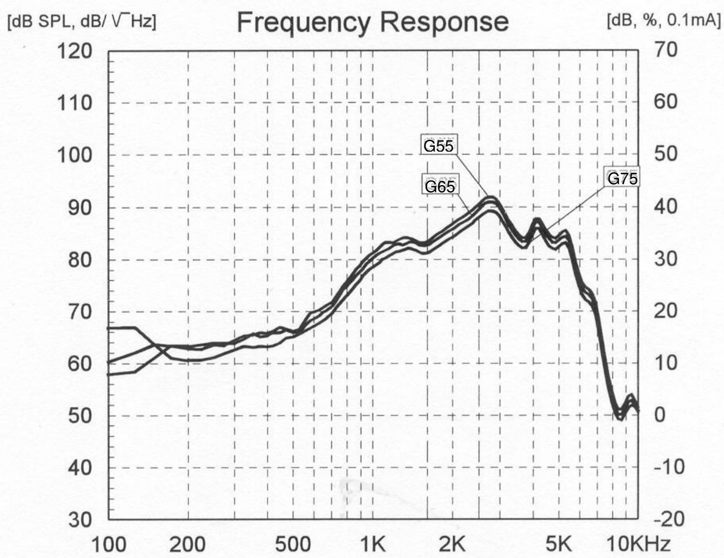
Hearing Instrument Test Results 12-07-06

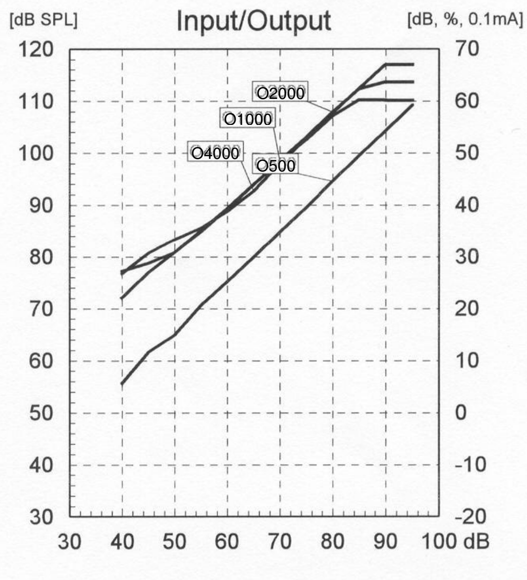
Non-standard test results

Client: LKH

ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** Right

Peak Gain 1: Input: 55 dB (WN)	42 dB at 2720 Hz
Peak Gain 2: Input: 65 dB (WN)	41 dB at 2720 Hz
Peak Gain 3: Input: 75 dB (WN)	39 dB at 2720 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3

D = Distortion [%] C = Current drain [mA]

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

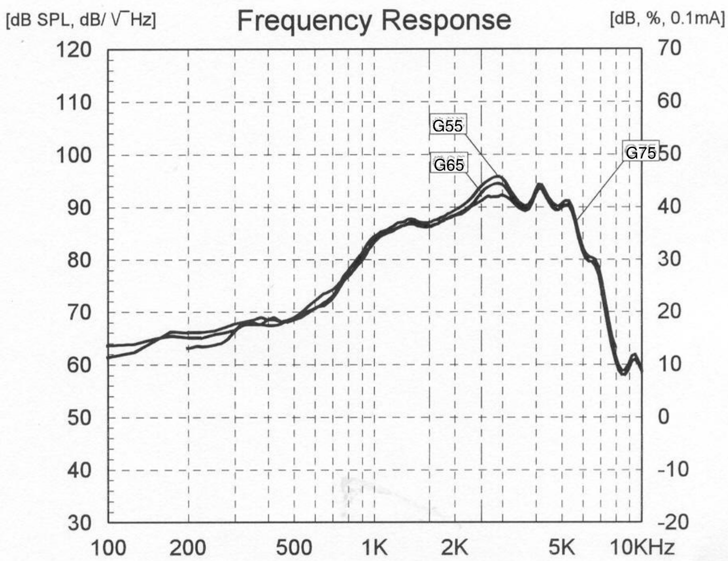
(Subject LKH, left hearing aid)

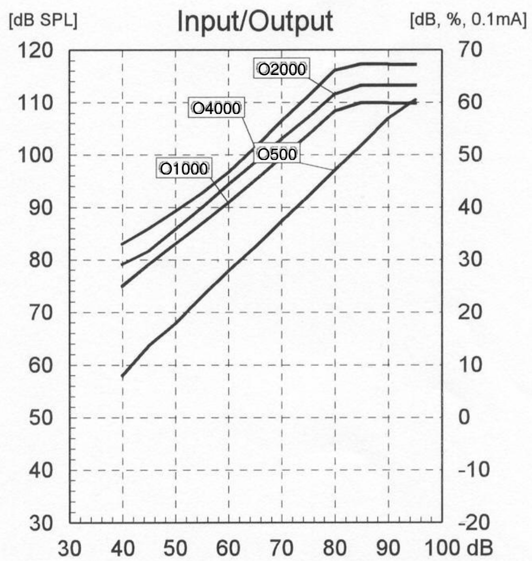
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: ~~BN~~ LKH
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** ~~Right~~ Left

Peak Gain 1: Input: 55 dB (WN)	46 dB at 2900 Hz
Peak Gain 2: Input: 65 dB (WN)	44 dB at 2900 Hz
Peak Gain 3: Input: 75 dB	44 dB at 4250 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

9.5.8. HA-data for subject JGH

Insertion gain-values adjusted to NAL-RP targets, are shown in the “General fine tuning data” (the parameters: *IGnormal*, *IGloud* and *IGsoft*)

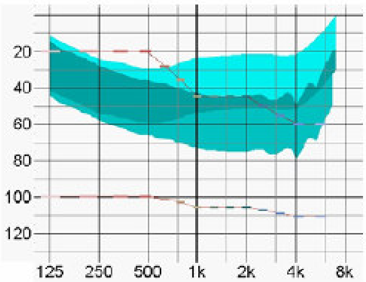
WIDEX
high definition hearing

COMPASS

Birth date:		Client no.:	0000017
Last name:		Print date:	12-07-2006
First name:		Printed by:	ABC

Right-ear hearing aid:	Diva SD-9M
Serial number:	170220
Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	20	45	45	60
Air-bone gap	---	---	---	---
Sensogram HTL	20	45	45	60
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling On
Fb Margin 6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	5	15	12	13
IGnormal	! 4	! 18	! 20	! 22
Target	-1	8	5	6
IGloud	! 4	! 18	! 20	! 21
Target	10	24	23	30
IGsoft	! 4	! 18	! 20	! 23

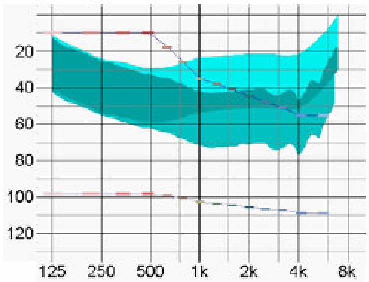
Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Left-ear hearing aid:	Diva SD-9M
Serial number:	170204
Fitting rationale:	Adult/Child > 5 years

Speech Spectra (normal)



Fitting data

	500Hz	1000Hz	2000Hz	4000Hz
Audiogram HTL	10	35	45	55
Air-bone gap	---	---	---	---
Sensogram HTL	10	35	45	55
Fb Test	---	---	---	---
Available Gain	---	---	---	---

Fb Cancelling On
Fb Margin 6

General fine tuning data

	500Hz	1000Hz	2000Hz	4000Hz
Target	1	12	11	10
IGnormal	! 0	! 16	! 17	! 20
Target	-5	5	4	3
IGloud	! 0	! 16	! 17	! 18
Target	4	19	23	27
IGsoft	! 0	! 17	! 18	! 21

Options

Occlusion Manager	Off
LF1	
LF2	
PRG	1: M M+T Tele MUS
Music program gain	0
LCT	OMNI
AOC	On
Telegain	0

AIS

Compass V3.4.1 - 1129 - Danmark

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

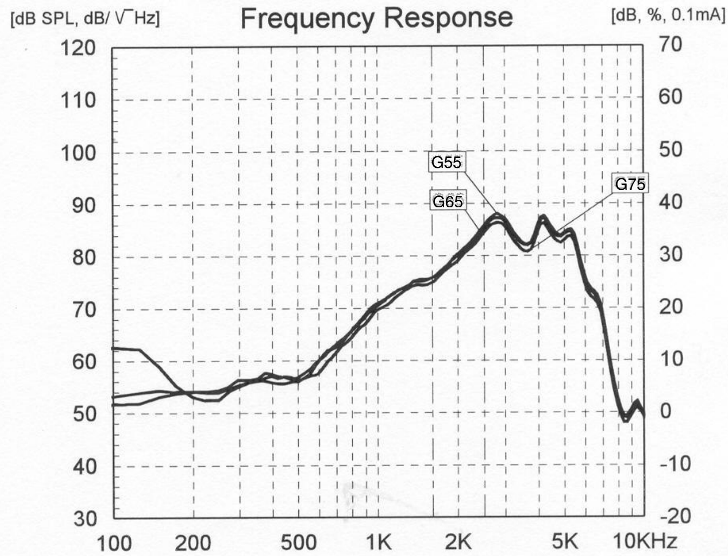
(Subject JGH, right hearing aid)

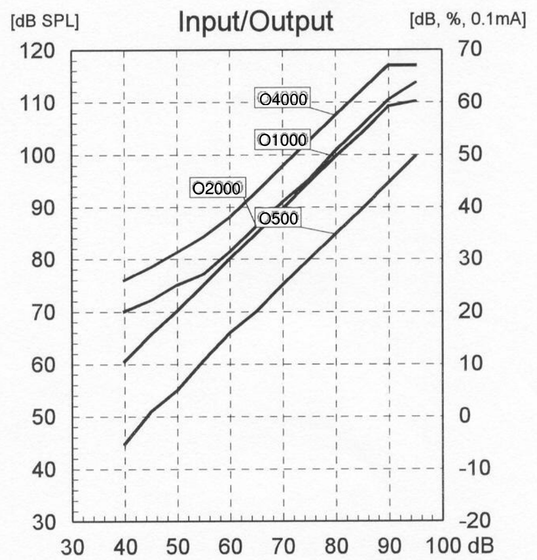
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: ~~BN~~ JGH
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** Right

Peak Gain 1: Input: 55 dB (WN)	38 dB at 2800 Hz
Peak Gain 2: Input: 65 dB (WN)	37 dB at 2800 Hz
Peak Gain 3: Input: 75 dB (WN)	36 dB at 2800 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

FFT-response (broadband noise) and input-output characteristics (500, 1000, 2000, 4000 Hz)

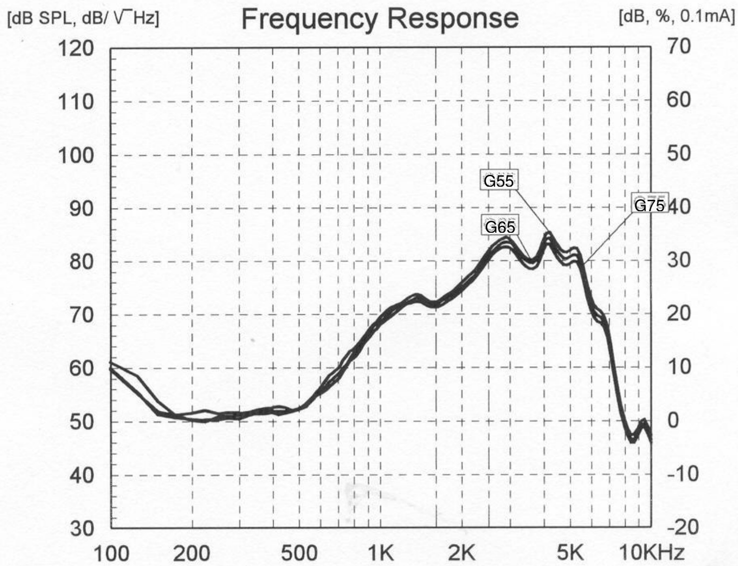
(Subject JGH, left hearing aid)

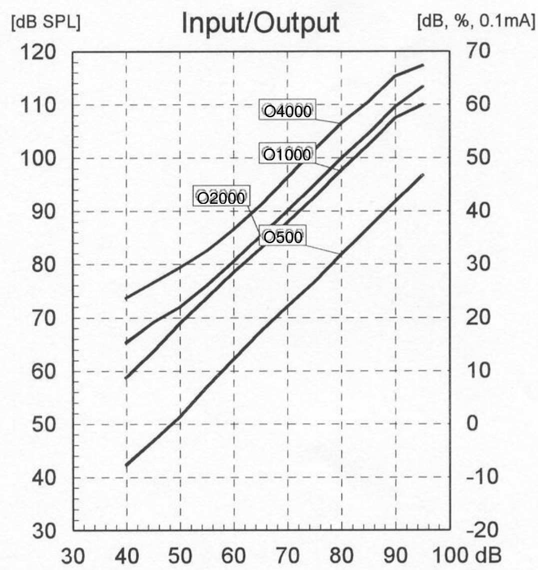
Hearing Instrument Test Results 12-07-06

Non-standard test results

Client: ~~BN~~ JGH
ClientNo: 0000066 **Hearing Aid:** AGC BTE **Ear:** ~~Right~~ Left.

Peak Gain 1: Input: 55 dB (WN)	35 dB at 4250 Hz
Peak Gain 2: Input: 65 dB (WN)	34 dB at 4250 Hz
Peak Gain 3: Input: 75 dB (WN)	33 dB at 4120 Hz





O = Output [dB SPL] G = Gain [dB] 1, 2, 3
D = Distortion [%] C = Current drain [mA]

9.6 Questionnaire concerning the use of the volume control in different listening situations

Spørgeskema om dit høreapparat og brugen af det i forskellige lydsituationer.

På de følgende sider skal du besvare nogle spørgsmål omkring dit høreapparat og behovet for at stille på styrkekontrollen i forskellige situationer.

Navn: ________________________

Fødselsdato: ___________________

1. Hvor længe har du brugt høreapparat (ca. år)? ___________

2. Hvilken type høreapparater bruger du for øjeblikket (mærke og model)?

   ____

3. Anvender du normalt apparat på begge ører, eller kun på et øre ad gangen:

   Begge ører ___ Kun på det ene øre ___

4. Hvor meget anvender du dit høreapparat i løbet af en dag (sæt kryds):

   Hele dagen ___ Halvdelen af dagen ____ Kun i visse situationer ____

4. Har dine høreapparater en styrkekontrol (eller volumenkontrol), du selv kan regulere på:

   JA ___ NEJ ___

Hvis du <u>har</u> en styrkekontrol på dit apparat, skal du besvare spørgsmålene på side 2 og 3.

Hvis du <u>ikke har</u> en styrkekontrol på dit apparat, skal du gå videre til spørgsmålene på side 4 og 5.

1

Spørgsmål til brugere som <u>har</u> styrkekontrol på deres høreapparat

På de følgende to sider er der 9 spørgsmål som omhandler brugen af styrkekontrollen i dit høreapparat, i forskellige lyttesituationer.

Hvert spørgsmål har to dele. Først skal du med et kryds markere, hvor ofte du stiller på styrkekontrollen i den beskrevne situation. Som vist i boksen nedenfor svarer A til *altid*, B til *næsten altid*, C til *hyppigt*, D til *halvdelen af tiden*, E til *indimellem*, F til *sjældent* og G svarer til *aldrig*.

Herefter skal du med et kryds markere, om du oftest skruer op eller ned for styrkekontrollen, **første gang du regulerer på den** i den givne situation.

Prøv at sætte krydserne ved de udsagn, der kommer tættest på din egen oplevelse i hverdagen. Hvis du ikke har oplevet den situation der beskrives i spørgsmålet, så prøv at forestille dig en lignende situation du har været i og giv et svar, der passer på denne. Genkender du slet ikke situationen, skal du bare springe spørgsmålet over.

A	Altid
B	Næsten altid
C	Hyppigt
D	Halvdelen af tiden
E	Indimellem
F	Sjældent
G	Aldrig

1.	Hvor ofte stiller du på apparatets styrkekontrol, når du er **til et selskab med flere mennesker der taler sammen**?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____
2.	Hvor ofte stiller du på apparatets styrkekontrol, når du skal **føre en samtale med en person på en restaurant eller en cafe**?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____
3.	Hvor ofte stiller du på apparatets styrkekontrol, når du **befinder dig ude i trafikken** (fx på en befærdet gade)?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____
4.	Hvor ofte stiller du på apparatets styrkekontrol, når du **kører med bus eller tog**?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____
5.	Hvor ofte stiller du på apparatets styrkekontrol, når du **er ude og handle i et supermarked eller storcenter**?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____

6.	Hvor ofte stiller du på apparatets styrkekontrol, når du **er til en koncert, et sportsstævne eller en anden begivenhed med mange mennesker**?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____
7.	Hvor ofte stiller du på apparatets styrkekontrol, når du **er i biografen og ser en film**?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____
8.	Hvor ofte stiller du på apparatets styrkekontrol, når du **lytter til radio eller TV derhjemme**?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____
9.	Hvor ofte stiller du på apparatets styrkekontrol, når du **kører bil og skal lytte til bilradioen**?	A B C D E F G
	- Når du justerer styrkekontrollen første gang i denne situation, skruer du da oftest op eller ned for styrken?	Op ____ Ned ____

10. Kan du nævne andre situationer, hvor du har brug for at stille på styrkekontrollen i dit høreapparat…

Situationer hvor du skruer op (nævn stikord):

____

____

____

Situationer hvor du skruer ned (nævn stikord):

____

____

____

11. Er der situationer, hvor du helt slukker for dit høreapparat (nævn stikord):

____

____

____

Tak for hjælpen!

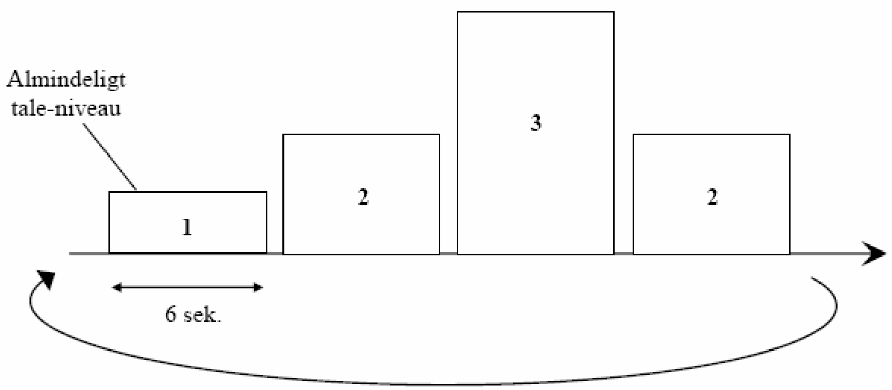
9.7 Written instruction provided to subjects in experiment #1

Oplevelsen af variation i lydens styrke

Til daglig lytter vi til lyde ved mange forskellige styrke-niveauer - lige fra helt svage lyde til mere kraftige lyde. Denne variation i lydens styrke vil vi gerne prøve at gengive i høreapparatet. I dette lytteforsøg fokuserer vi på oplevelsen af tale og baggrundsstøj ved kraftige styrke-niveauer.

Du skal nu lytte til en række optagelser af tale i to typer af baggrundsstøj. I hver optagelse indgår tre små del-optagelser. De tre små optagelser adskiller sig fra hinanden ved at være i forskellige styrke-niveauer. Den første del-optagelse (niveau 1) svarer altid til tale ved almindelig stemmestyrke, mens de to øvrige del-optagelser (niveau 2 og 3) gengiver to kraftigere niveauer.

På tegningen nedenfor er vist forløbet for en optagelse. Først præsenteres niveau 1 fulgt af niveau 2 og 3. Efter præsentationen af niveau 3 hører du igen niveau 2, hvorefter optagelsen kører forfra.



Prøv at lytte til to eksempler på hvordan optagelsen kan lyde. I det første eksempel er den oprindelige variation mellem de tre niveauer bevaret, mens lyden i det andet eksempel er behandlet sådan at der er en meget lille forskel mellem niveauerne.

(To lydeksempler)

Du skal nu skulle lytte til en række optagelser, hvor variationen mellem de tre lydstyrke-niveauer er forskellig fra optagelse til optagelse. Din opgave er at markere din oplevelse af niveau-variationerne på tre skalaer (se beskrivelsen på næste side).

Optagelserne kører i ring som vist på tegningen oven for. Du kan lytte så længe du vil til hver optagelse, førend du sætter dit kryds på skalaerne. Når du har sat dine tre krydser, går vi videre til den næste optagelse.

Skulle det ske, at en lyd bliver ubehagelig kraftig, skal du sige til med det samme. Lyden vil så blive afbrudt og du kan så foretage dine markeringer på skalaerne, hvorefter vi går videre til den næste optagelse.

Husk på, at der er ingen rigtige eller forkerte svar i dette forsøg. Din opgave er udelukkende at svare efter din bedste overbevisning ««««.

Beskrivelse af skalaerne.

På den første skala skal du markere, hvordan du oplever graden af variation mellem de tre niveauer…

- Hvis du synes der er en **stor variation** eller **meget stor variation** mellem de tre niveauer, skal du sætte dit kryds i højre side af skalaen (fra 6-10).
- Hvis du synes der er **ingen variation** eller kun **lille variation** mellem de tre niveauer, skal du sætte dit kryds i venstre side af skalaen (fra 0-4).
- Hvis du synes at variationen mellem de tre niveauer er **middel** (midt imellem lille og stor), skal du sætte et kryds i midten af skalaen (omkring 5).

Ingen variation | Lille variation | Middel | Stor variation | Meget stor variation

0 1 2 3 4 5 6 7 8 9 10

På den anden skala skal du markere, hvordan du oplever lydstyrken af de tre enkelte niveauer. Hvis du fx synes at **niveau 1** er komfortabel i lydstyrke, skal du sætte et 1-tal omkring 5 på skalaen. Hvis du synes at **niveau 2** fx ligger imellem komfortabel og kraftig, sætter du et 2-tal omkring 6 på skalaen, og hvis du synes at **niveau 3** fx ligger er meget kraftigt, sætter du et 3-tal omkring 9 på skalaen…

Ikke hørbar | Meget svag | Svag | Komfortabel | Kraftig | Meget kraftig | *Ubehagelig kraftig*

0 1 2 3 4 5 6 7 8 9 10

Udgangspunktet er som sagt, at vi gerne vil gengive variationer i lydens styrke gennem høreapparatet, uden at det bliver ubehageligt at lytte på. På den tredje skala skal du markere, hvor acceptabel du synes at gengivelsen af de tre niveauer ville være, hvis de forekom i en virkelig lyttesituation…

- Hvis du synes at **gengivelsen af de tre niveauer er acceptabel**, skal du sætte dit kryds til højre på skalaen (fra 6-10).
- Hvis du synes at **gengivelsen af de tre niveauer er uacceptabel**, skal du sætte dit kryds til venstre på skalaen (fra 0-4).
- Hvis du synes at **gengivelsen af de tre niveauer er nogenlunde**, skal du sætte dit kryds midt på skalaen (omkring 5).

Højst uacceptabel | Uacceptabel | Nogenlunde | Acceptabel | Højst acceptabel

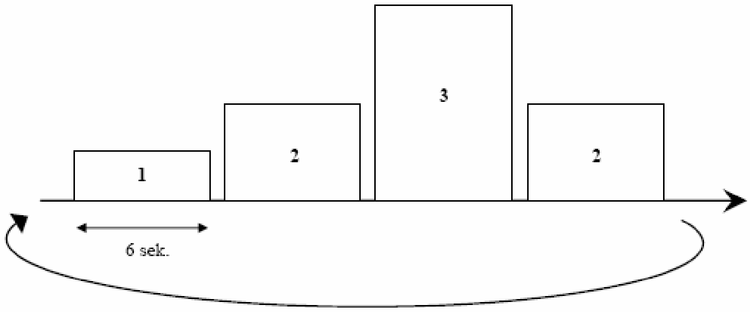
0 1 2 3 4 5 6 7 8 9 10

9.8 Sheet with categorical scales used in experiment #1

Navn: x	Dato: x	Kl. x	Test: x

Oplevelsen af variation i lydens styrke

Nr. …..



Hvordan oplever du graden af niveau-variation i denne optagelse?

Ingen variation — Lille variation — Middel — Stor variation — Meget stor variation

0 1 2 3 4 5 6 7 8 9 10

Marker med tallene 1, 2 og 3, hvordan du oplever lydstyrken af de tre niveauer:

Ikke hørbar — Meget svag — Svag — Komfortabel — Kraftig — Meget kraftig — *Ubehagelig kraftig*

0 1 2 3 4 5 6 7 8 9 10

Hvor acceptabel synes du at gengivelsen af de tre niveauer ville være, hvis de forekom i en virkelig lyttesituation?

Højst uacceptabel — Uacceptabel — Nogenlunde — Acceptabel — Højst acceptabel

0 1 2 3 4 5 6 7 8 9 10

9.9 Sheet with categorical scales used in experiment #2

Navn: x	Dato: x	Kl. x	Test: L/N+P

Nr. …..

Hvordan oplever du lydstyrken i denne optagelse?

Ikke hørbar	Meget svag	Svag	Komfortabel	Kraftig	Meget kraftig	*Ubehagelig kraftig*

0 1 2 3 4 5 6 7 8 9 10

Hvordan oplever du talens tydelighed i denne optagelse?

Meget utydelig	Utydelig	Nogenlunde	Tydelig	Meget tydelig

0 1 2 3 4 5 6 7 8 9 10

Hvordan oplever du baggrundsstøjen i denne optagelse?

Ikke hørbar	Ikke generende	Middel	Generende	Meget generende

0 1 2 3 4 5 6 7 8 9 10

Hvor acceptabel synes du at denne gengivelse ville være, hvis den forekom i en virkelig lyttesituation?

Højst uacceptabel	Uacceptabel	Nogenlunde	Acceptabel	Højst acceptabel

0 1 2 3 4 5 6 7 8 9 10

9.10 Data output from ANOVA's and model verifications in experiment #1

9.10.1 VARIATION SCALE

Model Dimension(b)				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	TRIAL	3		2
	CR	7		6
	SIGNAL	4		3
Random Effects	SUBJECT(a)	8	Variance Components	1
Residual				1
Total		23		14
a As of version 11.5, the syntax rules for the RANDOM subcommand have changed. Your command syntax may yield results that differ from those produced by prior versions. If you are using SPSS 11 syntax, please consult the current syntax reference guide for more information.				
b Dependent Variable: RATING.				

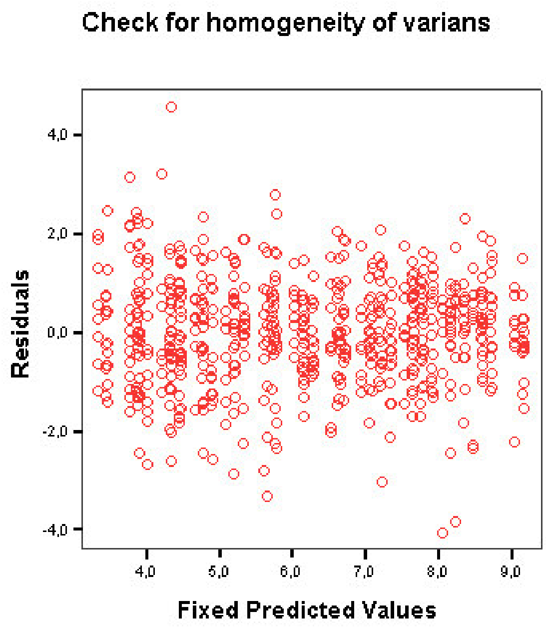
Information Criteria(a)	
-2 Restricted Log Likelihood	2041,974
Akaike's Information Criterion (AIC)	2045,974
Hurvich and Tsai's Criterion (AICC)	2045,993
Bozdogan's Criterion (CAIC)	2056,959
Schwarz's Bayesian Criterion (BIC)	2054,959
The information criteria are displayed in smaller-is-better forms.	
a Dependent Variable: RATING.	

Fixed Effects

Type III Tests of Fixed Effects(a)				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7,000	409,820	,000
TRIAL	2	653	,879	,416
CR	6	653	264,063	,000
SIGNAL	3	653	24,859	,000
a Dependent Variable: RATING.				

Covariance Parameters

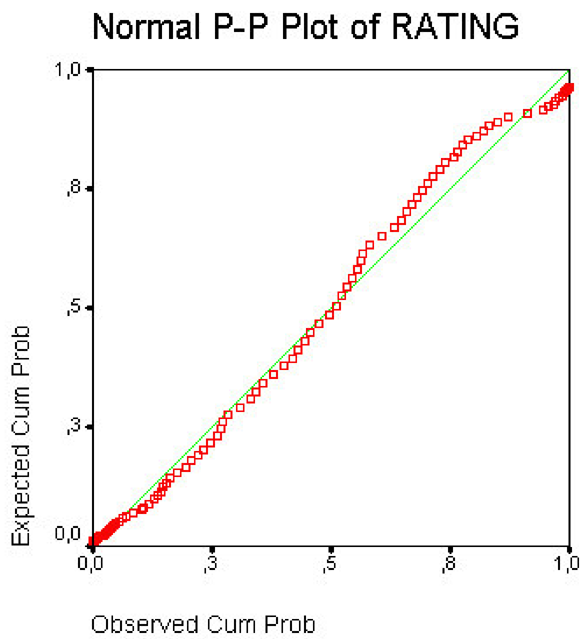
Estimates of Covariance Parameters(a)			
Parameter		Estimate	Std. Error
Residual		1,1379305	,0629759
SUBJECT	Variance	,7298242	,3973492
a Dependent Variable: RATING.			



PPlot

```
MODEL:   MOD_5.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean

For variable RATING ...
Normal distribution parameters estimated: location = 6,1709821 and scale =
2,1389665
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S - S C A L E (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	1731,1721	671	2,5800		
Within People	7632,1950	672	11,3574		
Between Measures	5845,4229	1	5845,4229	2195,1758	,0000
Residual	1786,7721	671	2,6628		
Nonadditivity	992,7654	1	992,7654	837,7170	,0000
Balance	794,0066	670	1,1851		
Total	9363,3671	1343	6,9720		
Grand Mean	4,0855				

Tukey estimate of power to which observations
must be raised to achieve additivity = -,4835

Reliability Coefficients

N of Cases = 672,0   N of Items = 2

Alpha = -,0321

9.10.2 LOUDNESS SCALE (1st segment)

Model Dimension(b)				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	TRIAL	3		2
	CR	7		6
	SIGNAL	4		3
Random Effects	SUBJECT(a)	8	Variance Components	1
Residual				1
Total		23		14
a As of version 11.5, the syntax rules for the RANDOM subcommand have changed. Your command syntax may yield results that differ from those produced by prior versions. If you are using SPSS 11 syntax, please consult the current syntax reference guide for more information.				
b Dependent Variable: RATING.				

Information Criteria(a)	
-2 Restricted Log Likelihood	1151,447
Akaike's Information Criterion (AIC)	1155,447
Hurvich and Tsai's Criterion (AICC)	1155,465
Bozdogan's Criterion (CAIC)	1166,431
Schwarz's Bayesian Criterion (BIC)	1164,431
The information criteria are displayed in smaller-is-better forms.	
a Dependent Variable: RATING.	

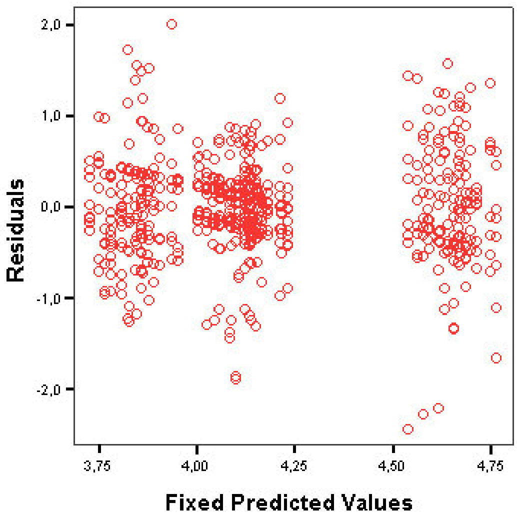
Fixed Effects

Type III Tests of Fixed Effects(a)				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7,000	221,836	,000
TRIAL	2	653	1,971	,140
CR	6	653	,598	,732
SIGNAL	3	653	66,112	,000
a Dependent Variable: RATING.				

Covariance Parameters

Estimates of Covariance Parameters(a)			
Parameter		Estimate	Std. Error
Residual		,2914847	,0161315
SUBJECT	Variance	,6267826	,3368843
a Dependent Variable: RATING.			

Check for homogeneity of varians

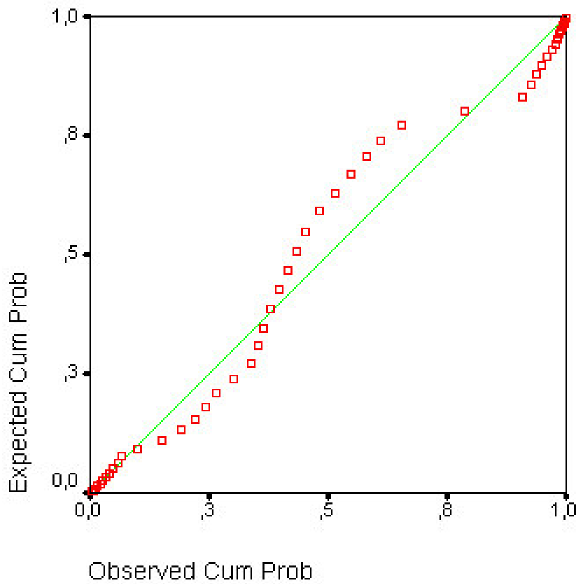


PPlot

```
MODEL:   MOD_6.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean

For variable RATING ...
Normal distribution parameters estimated: location = 4,180506 and scale =
,96197034
```

Normal P-P Plot of RATING



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S   -   S C A L E   (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	513,2673	671	,7649		
Within People	2153,2150	672	3,2042		
Between Measures	1597,5477	1	1597,5477	1929,1300	,0000
Residual	555,6673	671	,8281		
Nonadditivity	14,5667	1	14,5667	18,0367	,0000
Balance	541,1006	670	,8076		
Total	2666,4823	1343	1,9855		
Grand Mean	3,0903				

Tukey estimate of power to which observations
must be raised to achieve additivity = ,5225

Reliability Coefficients

N of Cases = 672,0 N of Items = 2

Alpha = -,0826

9.10.3 LOUDNESS SCALE (2nd segment)

Model Dimension(b)				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	TRIAL	3		2
	CR	7		6
	SIGNAL	4		3
Random Effects	SUBJECT(a)	8	Variance Components	1
Residual				1
Total		23		14
a As of version 11.5, the syntax rules for the RANDOM subcommand have changed. Your command syntax may yield results that differ from those produced by prior versions. If you are using SPSS 11 syntax, please consult the current syntax reference guide for more information.				
b Dependent Variable: RATING.				

Information Criteria(a)	
-2 Restricted Log Likelihood	1227,146
Akaike's Information Criterion (AIC)	1231,146
Hurvich and Tsai's Criterion (AICC)	1231,164
Bozdogan's Criterion (CAIC)	1242,130
Schwarz's Bayesian Criterion (BIC)	1240,130
The information criteria are displayed in smaller-is-better forms.	
a Dependent Variable: RATING.	

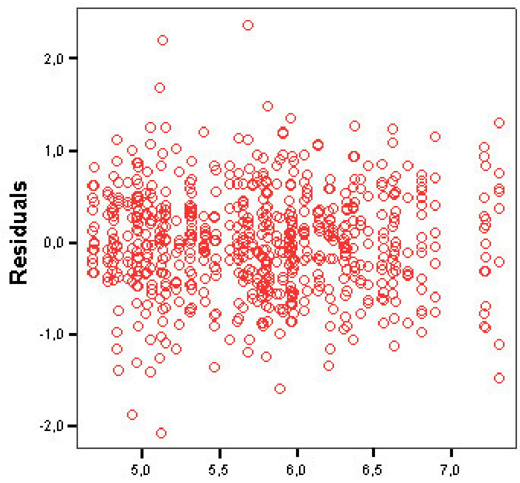
Fixed Effects

Type III Tests of Fixed Effects(a)				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7	436,693	,000
TRIAL	2	653,000	1,861	,156
CR	6	653,000	92,570	,000
SIGNAL	3	653,000	100,534	,000
a Dependent Variable: RATING.				

Covariance Parameters

Estimates of Covariance Parameters(a)			
Parameter		Estimate	Std. Error
Residual		,3274509	,0181219
SUBJECT	Variance	,6018223	,3237713
a Dependent Variable: RATING.			

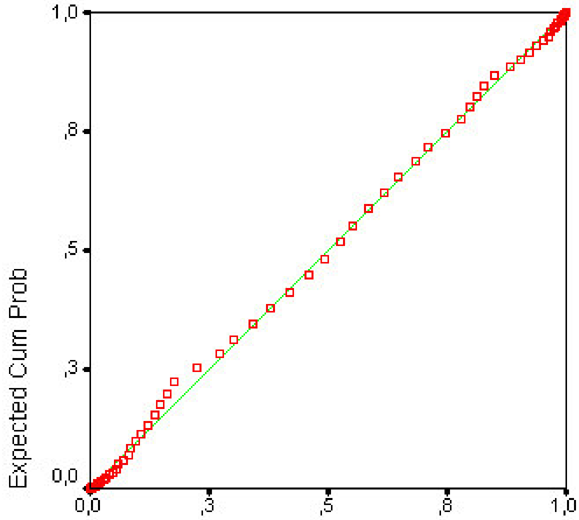
Check for homogeneity of varians



PPlot

```
MODEL:  MOD_7.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean

For variable RATING ...
Normal distribution parameters estimated: location = 5,7501488 and scale =
1,1267245
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S - S C A L E   (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	648,5200	671	,9665		
Within People	5376,6950	672	8,0010		
Between Measures	4725,3750	1	4725,3750	4868,1549	,0000
Residual	651,3200	671	,9707		
Nonadditivity	62,8688	1	62,8688	71,5813	,0000
Balance	588,4512	670	,8783		
Total	6025,2150	1343	4,4864		
Grand Mean	3,8751				

Tukey estimate of power to which observations
must be raised to achieve additivity = ,3565

Reliability Coefficients

N of Cases = 672,0   N of Items = 2

Alpha = -,0043

9.10.4 LOUDNESS SCALE (3rd segment)

Model Dimension(b)				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	TRIAL	3		2
	CR	7		6
	SIGNAL	4		3
Random Effects	SUBJECT(a)	8	Variance Components	1
Residual				1
Total		23		14
a As of version 11.5, the syntax rules for the RANDOM subcommand have changed. Your command syntax may yield results that differ from those produced by prior versions. If you are using SPSS 11 syntax, please consult the current syntax reference guide for more information.				
b Dependent Variable: RATING.				

Information Criteria(a)	
-2 Restricted Log Likelihood	1705,372
Akaike's Information Criterion (AIC)	1709,372
Hurvich and Tsai's Criterion (AICC)	1709,391
Bozdogan's Criterion (CAIC)	1720,357
Schwarz's Bayesian Criterion (BIC)	1718,357
The information criteria are displayed in smaller-is-better forms.	
a Dependent Variable: RATING.	

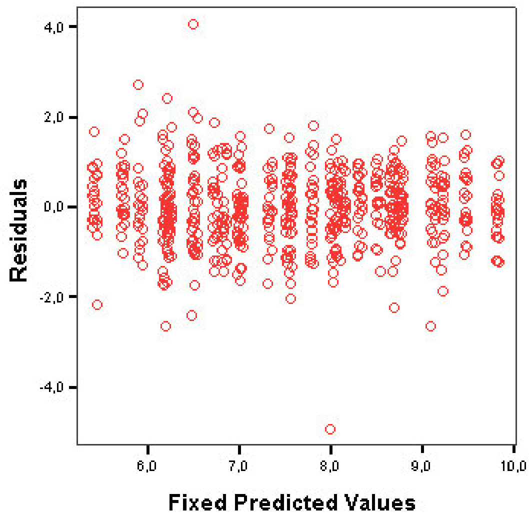
Fixed Effects

Type III Tests of Fixed Effects(a)				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7	534,073	,000
TRIAL	2	653,000	,138	,871
CR	6	653,000	211,234	,000
SIGNAL	3	653,000	53,296	,000
a Dependent Variable: RATING.				

Covariance Parameters

Estimates of Covariance Parameters(a)			
Parameter		Estimate	Std. Error
Residual		,6785990	,0375553
SUBJECT	Variance	,8442278	,4555771
a Dependent Variable: RATING.			

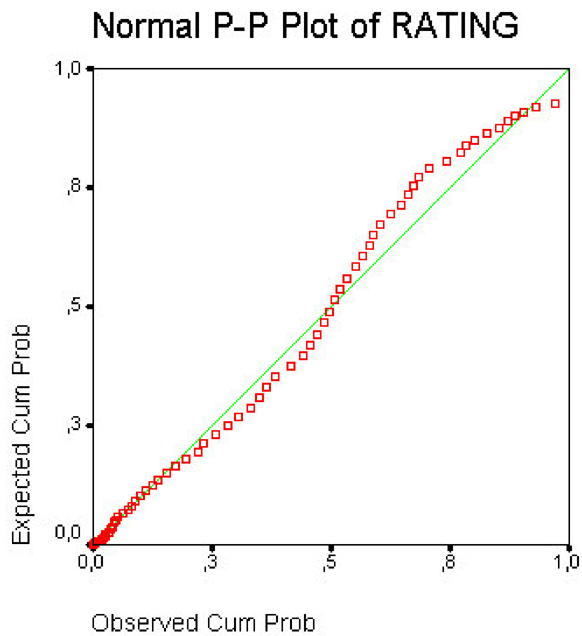
Check for homogeneity of varians



PPlot

```
MODEL:  MOD_8.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean

For variable RATING ...
Normal distribution parameters estimated: location = 7,5431548 and scale =
1,6884938
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S   -   S C A L E   (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	1176,8143	671	1,7538		
Within People	11508,3400	672	17,1255		
Between Measures	10324,1257	1	10324,1257	5849,8606	,0000
Residual	1184,2143	671	1,7648		
Nonadditivity	455,9574	1	455,9574	419,4831	,0000
Balance	728,2569	670	1,0870		
Total	12685,1543	1343	9,4454		
Grand Mean	4,7716				

Tukey estimate of power to which observations
must be raised to achieve additivity = -,0716

Reliability Coefficients

N of Cases = 672,0 N of Items = 2

Alpha = -,0063

9.10.5 ACCEPTANCE SCALE

Model Dimension(b)				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	TRIAL	3		2
	CR	7		6
	SIGNAL	4		3
Random Effects	SUBJECT(a)	8	Variance Components	1
Residual				1
Total		23		14

a As of version 11.5, the syntax rules for the RANDOM subcommand have changed. Your command syntax may yield results that differ from those produced by prior versions. If you are using SPSS 11 syntax, please consult the current syntax reference guide for more information.

b Dependent Variable: RATING.

Information Criteria(a)	
-2 Restricted Log Likelihood	2427,498
Akaike's Information Criterion (AIC)	2431,498
Hurvich and Tsai's Criterion (AICC)	2431,516
Bozdogan's Criterion (CAIC)	2442,483
Schwarz's Bayesian Criterion (BIC)	2440,483

The information criteria are displayed in smaller-is-better forms.

a Dependent Variable: RATING.

Fixed Effects

Type III Tests of Fixed Effects(a)				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	7,000	411,874	,000
TRIAL	2	653	,060	,942
CR	6	653	61,353	,000
SIGNAL	3	653	27,120	,000

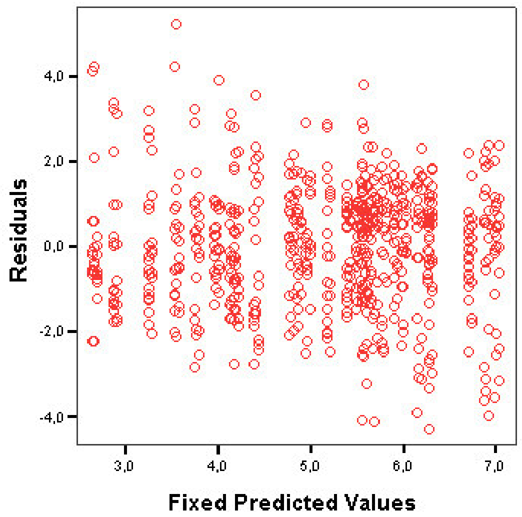
a Dependent Variable: RATING.

Covariance Parameters

Estimates of Covariance Parameters(a)			
Parameter		Estimate	Std. Error
Residual		2,0620224	,1141174
SUBJECT	Variance	,4831643	,2713870

a Dependent Variable: RATING.

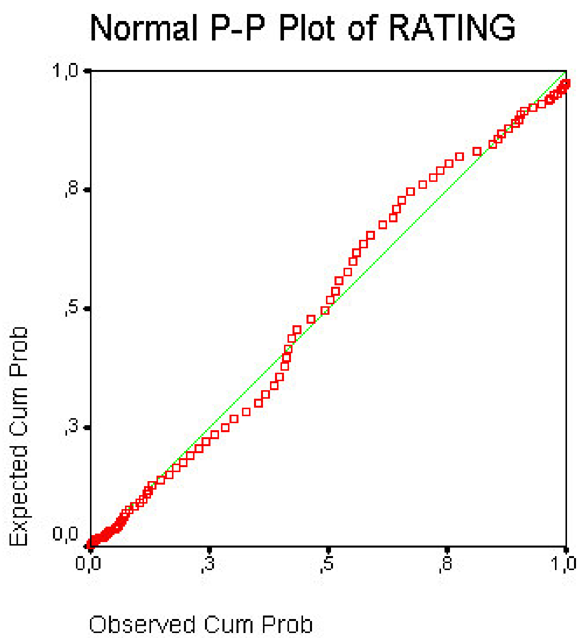
Check for homogeneity of varians



PPlot

```
MODEL:  MOD_9.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean

For variable RATING ...
Normal distribution parameters estimated: location = 5,1126488 and scale =
1,9578732
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S   -   S C A L E   (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	1502,5612	671	2,2393		
Within People	4772,9250	672	7,1026		
Between Measures	3255,3638	1	3255,3638	1439,3812	,0000
Residual	1517,5612	671	2,2616		
Nonadditivity	750,7009	1	750,7009	655,8816	,0000
Balance	766,8603	670	1,1446		
Total	6275,4862	1343	4,6727		
Grand Mean	3,5563				

Tukey estimate of power to which observations
must be raised to achieve additivity = -,6152

Reliability Coefficients

N of Cases = 672,0 N of Items = 2

Alpha = -,0100

9.11 Data output from ANOVA's and model verifications in experiment #2

9.11.1 Loud speech & party-noise: LOUDNESS SCALE

Mixed Model Analysis, L+P, Loudness

Model Dimension				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	COMP	16		15
	TRIAL	3		2
Random Effects	SUBJECT	7	Variance Components	1
Residual				1
Total		27		20

Information Criteria	
-2 Restricted Log Likelihood	923,000
Akaike's Information Criterion (AIC)	927,000
Hurvich and Tsai's Criterion (AICC)	927,038
Bozdogan's Criterion (CAIC)	936,524
Schwarz's Bayesian Criterion (BIC)	934,524

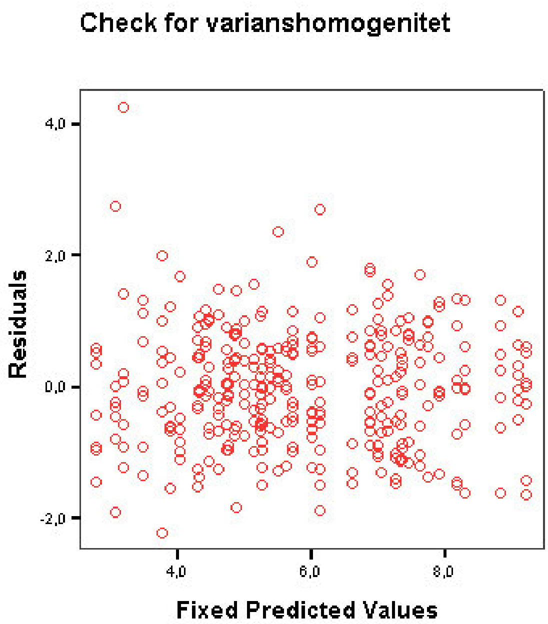
Fixed Effects

Type III Tests of Fixed Effects				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6	684,069	,000
COMP	15	312,000	70,800	,000
TRIAL	2	312,000	5,544	,004

Covariance Parameters

Estimates of Covariance Parameters			
Parameter		Estimate	Std. Error
Residual		,8425080	,0674546
SUBJECT	Variance	,3350343	,2035708

Interactive Graph

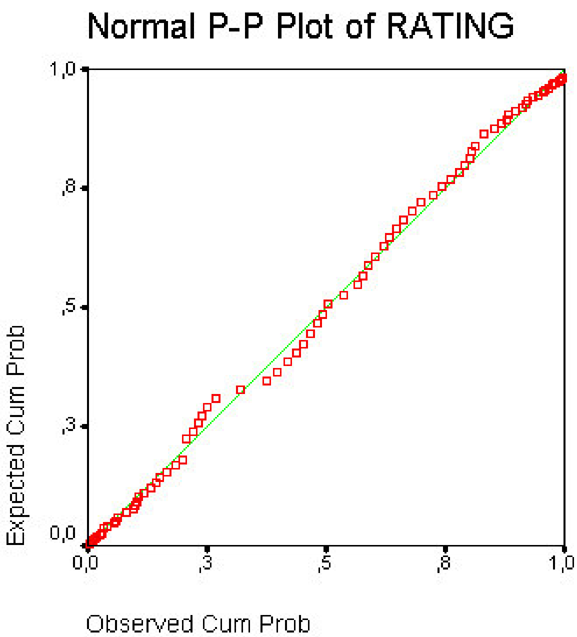


PPlot

```
MODEL:  MOD_1.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean



For variable RATING ...
Normal distribution parameters estimated: location = 5,8699405 and scale =
1,9458999
```



Reliability

```
****** Method 1 (space saver) will be used for this analysis ******



  R E L I A B I L I T Y   A N A L Y S I S   -   S C A L E   (A L P H A)



                       Analysis of Variance

Source of Variation     Sum of Sq.       DF       Mean Square      F
Prob.

Between People            701,6432       335           2,0945
Within People            3306,8850       336           9,8419
  Between Measures         2516,0418        1      2516,0418   1065,7916
,0000
  Residual                  790,8432      335          2,3607
    Nonadditivity            388,7132        1      388,7132    322,8563
,0000
    Balance                  402,1300      334          1,2040
Total                    4008,5282       671           5,9740
     Grand Mean        3,9350


Tukey estimate of power to which observations
must be raised to achieve additivity         =      -,5136



Reliability Coefficients

N of Cases =    336,0                    N of Items =  2

Alpha =   -,1271
```

9.11.2 Loud speech & party-noise: SPEECH CLEARNESS SCALE

Mixed Model Analysis, L+P, Speech

Model Dimension				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	COMP	16		15
	TRIAL	3		2
Random Effects	SUBJECT	7	Variance Components	1
Residual				1
Total		27		20

Information Criteria	
-2 Restricted Log Likelihood	961,387
Akaike's Information Criterion (AIC)	965,387
Hurvich and Tsai's Criterion (AICC)	965,425
Bozdogan's Criterion (CAIC)	974,911
Schwarz's Bayesian Criterion (BIC)	972,911

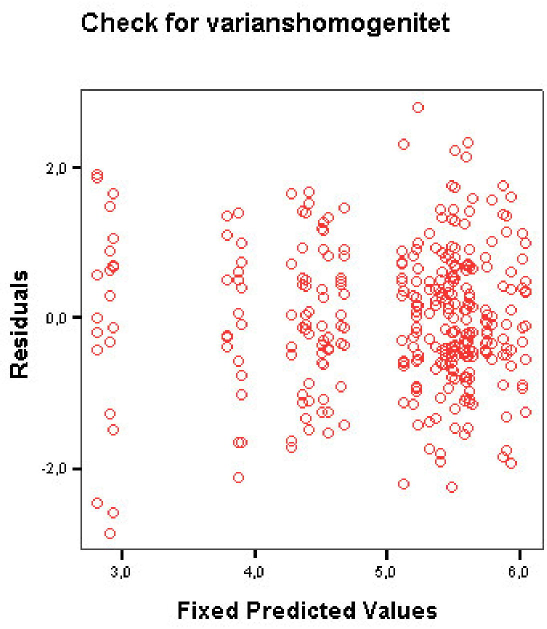
Fixed Effects

Type III Tests of Fixed Effects				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6	196,153	,000
COMP	15	312	15,537	,000
TRIAL	2	312	,425	,654

Covariance Parameters

Estimates of Covariance Parameters			
Parameter		Estimate	Std. Error
Residual		,9354430	,0748954
SUBJECT	Variance	,8983454	,5299139

Interactive Graph

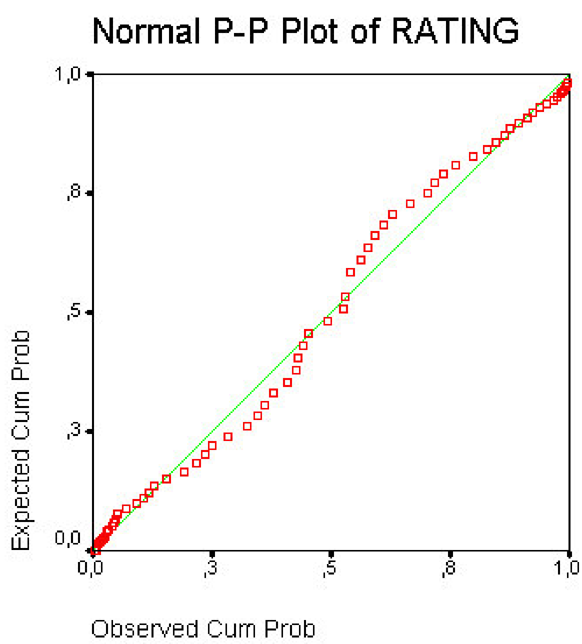


PPlot

```
MODEL:  MOD_2.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean



For variable RATING ...
Normal distribution parameters estimated: location = 5,0714286 and scale =
1,5210029
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S   -   S C A L E   (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	509,8029	335	1,5218		
Within People	2074,0600	336	6,1728		
Between Measures	1584,8571	1	1584,8571	1085,2904	,0000
Residual	489,2029	335	1,4603		
Nonadditivity	148,8847	1	148,8847	146,1205	,0000
Balance	340,3182	334	1,0189		
Total	2583,8629	671	3,8508		
Grand Mean	3,5357				

Tukey estimate of power to which observations
must be raised to achieve additivity = -,2442

Reliability Coefficients

N of Cases = 336,0   N of Items = 2

Alpha = ,0404

9.11.3 Loud speech & party-noise: NOISINESS SCALE

Mixed Model Analysis, L+P, Noise

Model Dimension				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	COMP	16		15
	TRIAL	3		2
Random Effects	SUBJECT	7	Variance Components	1
Residual				1
Total		27		20

Information Criteria	
-2 Restricted Log Likelihood	955,301
Akaike's Information Criterion (AIC)	959,301
Hurvich and Tsai's Criterion (AICC)	959,339
Bozdogan's Criterion (CAIC)	968,825
Schwarz's Bayesian Criterion (BIC)	966,825

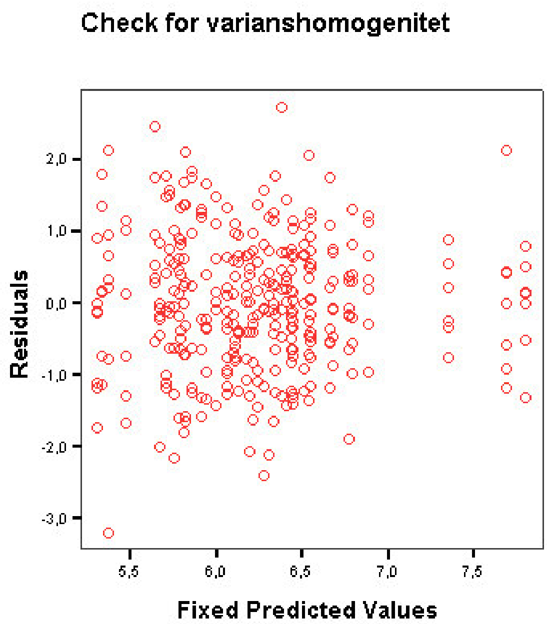
Fixed Effects

Type III Tests of Fixed Effects				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6	434,885	,000
COMP	15	312	5,929	,000
TRIAL	2	312	6,537	,002

Covariance Parameters

Estimates of Covariance Parameters			
Parameter		Estimate	Std. Error
Residual		,9242111	,0739961
SUBJECT	Variance	,6044840	,3601188

Interactive Graph

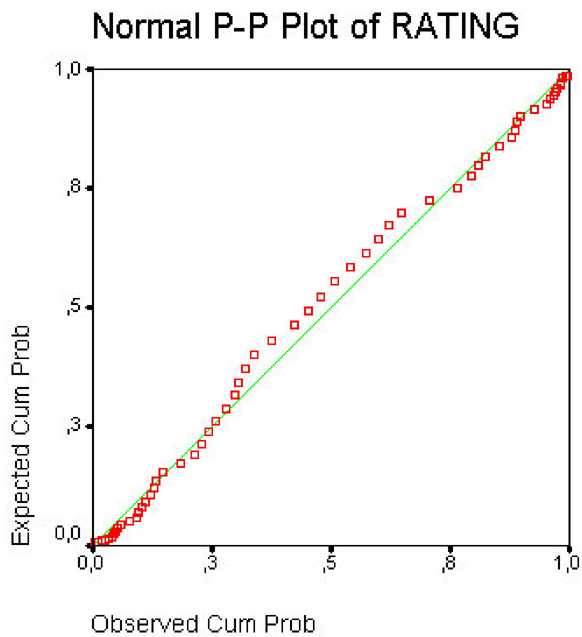


PPlot

```
MODEL:  MOD_3.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean



For variable RATING ...
Normal distribution parameters estimated: location = 6,225 and scale =
1,2955377
```



Reliability

```
****** Method 1 (space saver) will be used for this analysis ******



  R E L I A B I L I T Y   A N A L Y S I S  -  S C A L E  (A L P H A)


                       Analysis of Variance

Source of Variation     Sum of Sq.     DF      Mean Square      F
Prob.

Between People           343,2350      335          1,0246
Within People           3441,9400      336         10,2439
  Between Measures        2998,9050       1      2998,9050   2267,6158
,0000
  Residual                 443,0350     335          1,3225
    Nonadditivity           83,3442       1         83,3442     77,3914
,0000
    Balance                359,6908     334          1,0769
Total                   3785,1750      671          5,6411
     Grand Mean       4,1125


Tukey estimate of power to which observations
must be raised to achieve additivity        =      ,0407


Reliability Coefficients

N of Cases =    336,0                  N of Items =  2

Alpha =   -,2908
```

9.11.4 Loud speech & party-noise: ACCEPTANCE SCALE

Mixed Model Analysis, L+P, Acceptance

Model Dimension

		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	COMP	16		15
	TRIAL	3		2
Random Effects	SUBJECT	7	Variance Components	1
Residual				1
Total		27		20

Information Criteria

-2 Restricted Log Likelihood	934,505
Akaike's Information Criterion (AIC)	938,505
Hurvich and Tsai's Criterion (AICC)	938,544
Bozdogan's Criterion (CAIC)	948,030
Schwarz's Bayesian Criterion (BIC)	946,030

Fixed Effects

Type III Tests of Fixed Effects

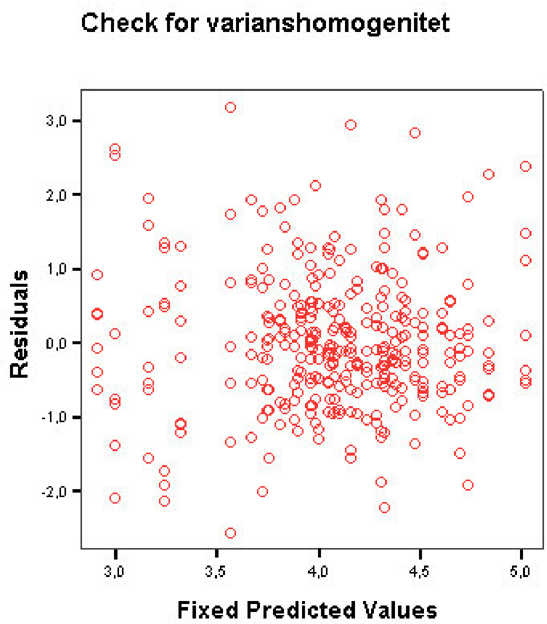
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6	847,084	,000
COMP	15	312	4,384	,000
TRIAL	2	312	5,806	,003

Covariance Parameters

Estimates of Covariance Parameters

Parameter		Estimate	Std. Error
Residual		,8901456	,0712687
SUBJECT	Variance	,1188387	,0793323

Interactive Graph

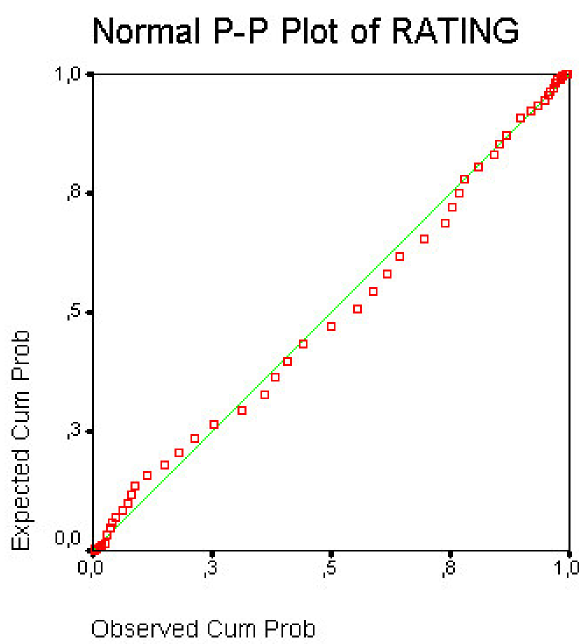


PPlot

```
MODEL:  MOD_4.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean



For variable RATING ...
Normal distribution parameters estimated: location = 4,077381 and scale =
1,0736576
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S - S C A L E (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	350,6840	335	1,0468		
Within People	984,4900	336	2,9300		
Between Measures	725,0060	1	725,0060	935,9997	,0000
Residual	259,4840	335	,7746		
Nonadditivity	18,7480	1	18,7480	26,0112	,0000
Balance	240,7361	334	,7208		
Total	1335,1740	671	1,9898		
Grand Mean	3,0387				

Tukey estimate of power to which observations
must be raised to achieve additivity = ,3236

Reliability Coefficients

N of Cases = 336,0   N of Items = 2

Alpha = ,2601

9.11.5 Normal speech & party-noise: LOUDNESS SCALE

Mixed Model Analysis, N+P, Loudness

Model Dimension				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	COMP	17		16
	TRIAL	3		2
Random Effects	SUBJECT	7	Variance Components	1
Residual				1
Total		28		21

Information Criteria	
-2 Restricted Log Likelihood	769,237
Akaike's Information Criterion (AIC)	773,237
Hurvich and Tsai's Criterion (AICC)	773,274
Bozdogan's Criterion (CAIC)	782,779
Schwarz's Bayesian Criterion (BIC)	780,779

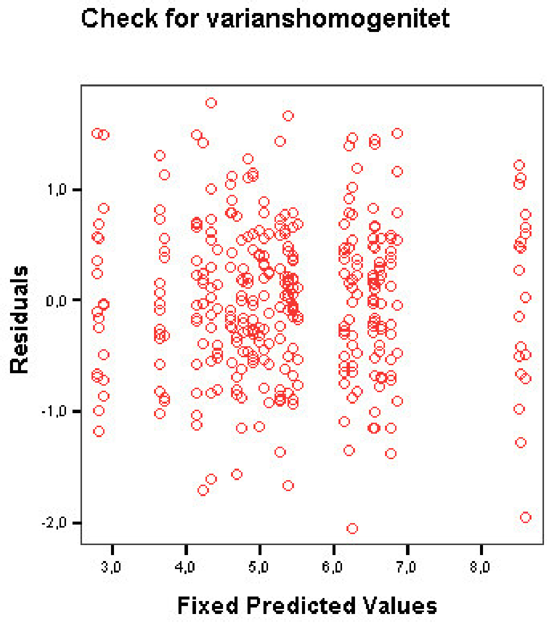
Fixed Effects

Type III Tests of Fixed Effects				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	6,012	481,292	,000
COMP	16	315,013	73,861	,000
TRIAL	2	315,503	,438	,646

Covariance Parameters

Estimates of Covariance Parameters			
Parameter		Estimate	Std. Error
Residual		,4989385	,0397556
SUBJECT	Variance	,4112756	,2433082

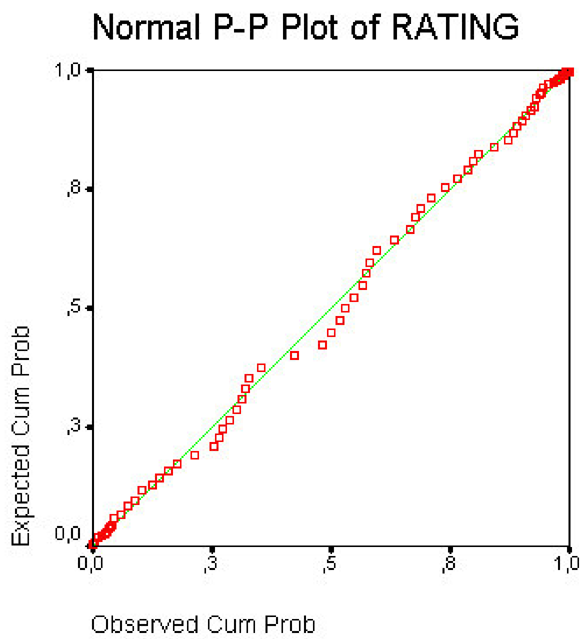
Interactive Graph



PPlot

```
MODEL:  MOD_5.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean


For variable RATING ...
Normal distribution parameters estimated: location = 5,4061765 and scale =
1,6046182
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S - S C A L E (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	555,1085	339	1,6375		
Within People	2568,5750	340	7,5546		
Between Measures	2030,6765	1	2030,6765	1279,7941	,0000
Residual	537,8985	339	1,5867		
Nonadditivity	191,8663	1	191,8663	187,4126	,0000
Balance	346,0322	338	1,0238		
Total	3123,6835	679	4,6004		
Grand Mean	3,6781				

Tukey estimate of power to which observations
must be raised to achieve additivity = -,2513

Reliability Coefficients

N of Cases = 340,0   N of Items = 2

Alpha = ,0310

9.11.6 Normal speech & party-noise: SPEECH CLEARNESS SCALE

Mixed Model Analysis, N+P, Speech

Model Dimension				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	COMP	17		16
	TRIAL	3		2
Random Effects	SUBJECT	7	Variance Components	1
Residual				1
Total		28		21

Information Criteria	
-2 Restricted Log Likelihood	866,773
Akaike's Information Criterion (AIC)	870,773
Hurvich and Tsai's Criterion (AICC)	870,810
Bozdogan's Criterion (CAIC)	880,315
Schwarz's Bayesian Criterion (BIC)	878,315

Fixed Effects

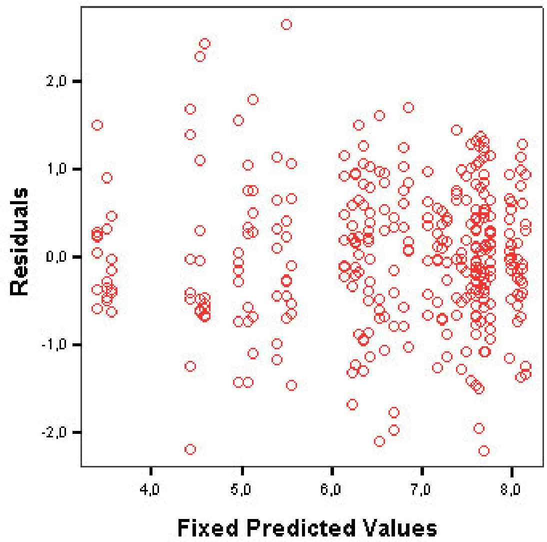
Type III Tests of Fixed Effects				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	5,966	699,770	,000
COMP	16	314,967	52,600	,000
TRIAL	2	315,594	,998	,370

Covariance Parameters

Estimates of Covariance Parameters			
Parameter		Estimate	Std. Error
Residual		,6793441	,0541343
SUBJECT	Variance	,4300209	,2573128

Interactive Graph

Check for varianshomogenitet

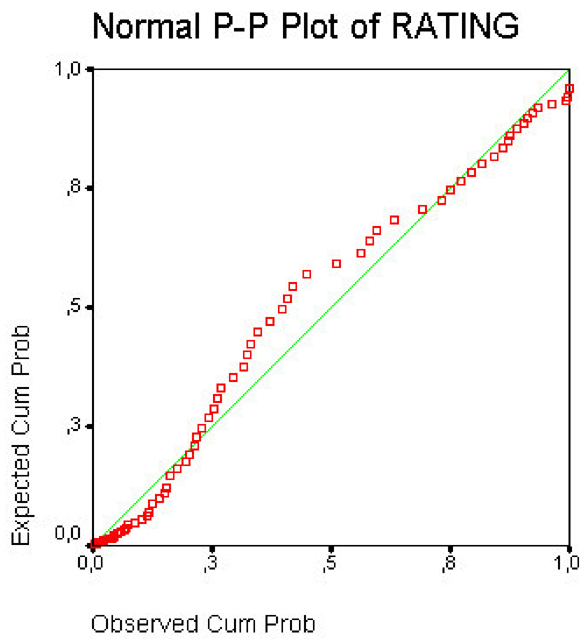


PPlot

```
MODEL:  MOD_6.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean



For variable RATING ...
Normal distribution parameters estimated: location = 6,6185294 and scale =
1,6355141
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S   -   S C A L E   (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	553,5866	339	1,6330		
Within People	4278,5350	340	12,5839		
Between Measures	3705,1784	1	3705,1784	2190,7054	,0000
Residual	573,3566	339	1,6913		
Nonadditivity	212,9201	1	212,9201	199,6663	,0000
Balance	360,4365	338	1,0664		
Total	4832,1216	679	7,1165		
Grand Mean	4,2843				

Tukey estimate of power to which observations
must be raised to achieve additivity = -,1383

Reliability Coefficients

N of Cases = 340,0   N of Items = 2

Alpha = -,0357

9.11.7 Normal speech & party-noise: NOSINESS SCALE

Mixed Model Analysis, N+P, Noise

Model Dimension				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	COMP	17		16
	TRIAL	3		2
Random Effects	SUBJECT	7	Variance Components	1
Residual				1
Total		28		21

Information Criteria	
-2 Restricted Log Likelihood	883,175
Akaike's Information Criterion (AIC)	887,175
Hurvich and Tsai's Criterion (AICC)	887,213
Bozdogan's Criterion (CAIC)	896,718
Schwarz's Bayesian Criterion (BIC)	894,718

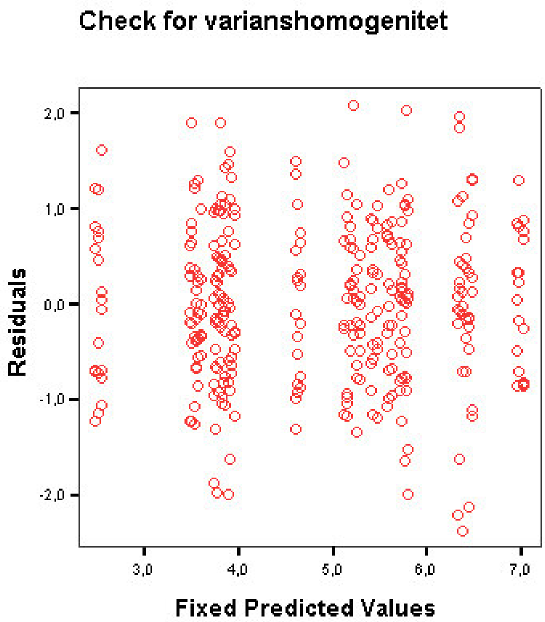
Fixed Effects

Type III Tests of Fixed Effects				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	5,987	162,266	,000
COMP	16	314,987	44,482	,000
TRIAL	2	315,284	,166	,847

Covariance Parameters

Estimates of Covariance Parameters			
Parameter		Estimate	Std. Error
Residual		,7045233	,0561389
SUBJECT	Variance	,9972462	,5850180

Interactive Graph

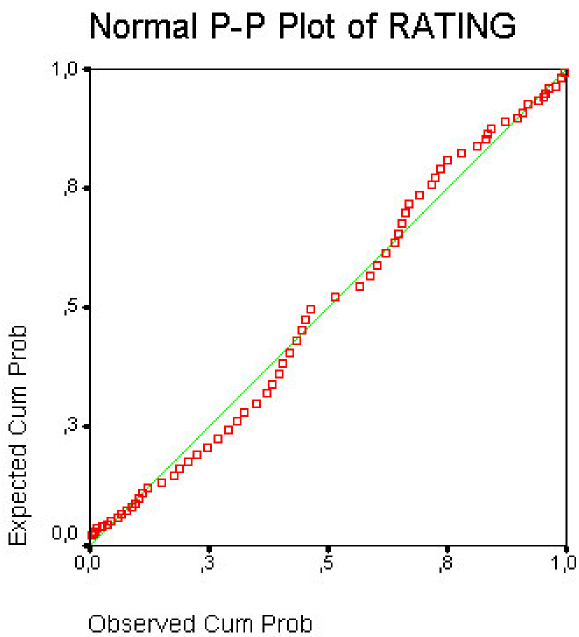


PPlot

```
MODEL:  MOD_7.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean



For variable RATING ...
Normal distribution parameters estimated: location = 4,9091176 and scale =
1,719489
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S   -   S C A L E   (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	626,8809	339	1,8492		
Within People	2084,1550	340	6,1299		
Between Measures	1488,5841	1	1488,5841	847,3047	,0000
Residual	595,5709	339	1,7568		
Nonadditivity	243,9703	1	243,9703	234,5331	,0000
Balance	351,6005	338	1,0402		
Total	2711,0359	679	3,9927		
Grand Mean	3,4296				

Tukey estimate of power to which observations
must be raised to achieve additivity = -,4460

Reliability Coefficients

N of Cases = 340,0   N of Items = 2

Alpha = ,0499

9.11.8 Normal speech & party-noise: ACCEPTANCE SCALE

Mixed Model Analysis, N+P, Acceptance

Model Dimension				
		Number of Levels	Covariance Structure	Number of Parameters
Fixed Effects	Intercept	1		1
	COMP	17		16
	TRIAL	3		2
Random Effects	SUBJECT	7	Variance Components	1
Residual				1
Total		28		21

Information Criteria	
-2 Restricted Log Likelihood	978,563
Akaike's Information Criterion (AIC)	982,563
Hurvich and Tsai's Criterion (AICC)	982,601
Bozdogan's Criterion (CAIC)	992,106
Schwarz's Bayesian Criterion (BIC)	990,106

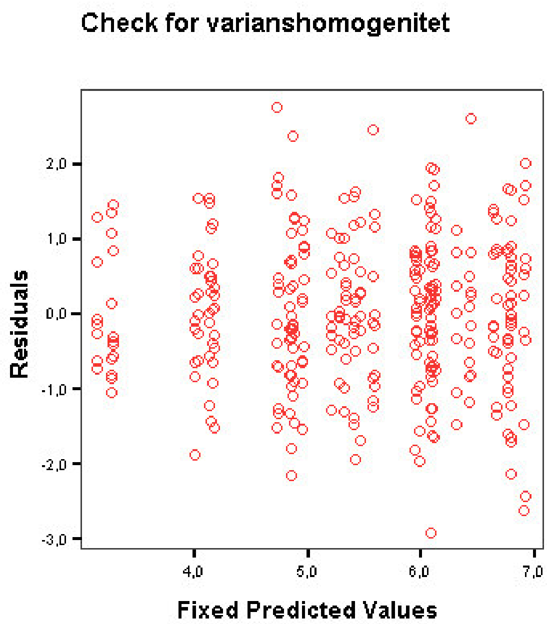
Fixed Effects

Type III Tests of Fixed Effects				
Source	Numerator df	Denominator df	F	Sig.
Intercept	1	5,938	443,951	,000
COMP	16	314,939	22,195	,000
TRIAL	2	315,758	,650	,522

Covariance Parameters

Estimates of Covariance Parameters			
Parameter		Estimate	Std. Error
Residual		,9675779	,0771059
SUBJECT	Variance	,4533819	,2750111

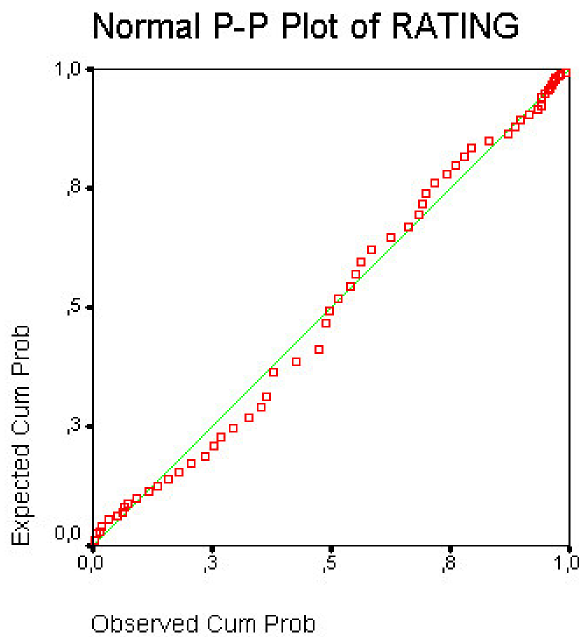
Interactive Graph



PPlot

```
MODEL:  MOD_8.
Distribution tested: Normal
Proportion estimation formula used: Blom's
Rank assigned to ties: Mean


For variable RATING ...
Normal distribution parameters estimated: location = 5,4314706 and scale =
1,5123223
```



Reliability

****** Method 1 (space saver) will be used for this analysis ******

R E L I A B I L I T Y   A N A L Y S I S   -   S C A L E   (A L P H A)

Analysis of Variance

Source of Variation	Sum of Sq.	DF	Mean Square	F	Prob.
Between People	480,7766	339	1,4182		
Within People	2575,2150	340	7,5742		
Between Measures	2060,5084	1	2060,5084	1357,1077	,0000
Residual	514,7066	339	1,5183		
Nonadditivity	160,2763	1	160,2763	152,8464	,0000
Balance	354,4303	338	1,0486		
Total	3055,9916	679	4,5007		
Grand Mean	3,6907				

Tukey estimate of power to which observations
must be raised to achieve additivity = -,2242

Reliability Coefficients

N of Cases = 340,0   N of Items = 2

Alpha = -,0706